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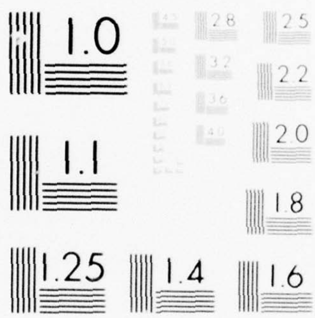

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RADC-TR-77-181
Phase Report
May 1977

DIGROUP DATA REDUCTION TECHNIQUES

North Carolina State University

ROME AIR DEVELOPMENT CENTER
Air Force Systems Command
Griffiss Air Force Base, New York 13441

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APPROVED: Jacob Scherer
JACOB SCHERER
Project Engineer

APPROVED: Joseph J. Naresky
JOSEPH J. NARESKY
Chief, Reliability & Compatibility Division

FOR THE COMMANDER: John P. Huss
JOHN P. HUSS
Acting Chief, Plans Office

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20. ABSTRACT (Continue on reverse side if necessary and identify by block number)

This report describes a study made to determine the best speech data compression system to use in compressing the output of two T1 carrier systems down into the transmission rate normally occupied by one T1 system. The study contains a review of previously proposed speech compression techniques, a more detailed consideration of three candidate systems for this application and the rationale for selecting ADPCM/TASI as the most suitable system. ADPCM/TASI is a combination of Adaptive Differential Pulse Code Modulation OVER

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(ADPCM) and Time Assignment Speech Interpolation (TASI). Computer simulation was the primary technique used to evaluate the systems.

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PREFACE

This effort was conducted by North Carolina State University under the sponsorship of the Rome Air Development Center Post-Doctoral Program. Capt. Sheldon Kelley of RADC was the task project engineer and provided overall technical direction and guidance. The authors of this report are Drs. J.B. O'Neal, Jr. and R.W. Stroh of NC State University.

The RADC Post-Doctoral Program is a cooperative venture between RADC and some sixty-five universities eligible to participate in the program. Syracuse University (Dept. of Electrical and Computer Engineering), Purdue University (School of Electrical Engineering), and State University of New York at Buffalo (Dept. of Electrical Engineering) act as prime contractor schools with other schools participating via sub-contracts with the prime schools. The U.S. Air Force Academy (Dept. of Electrical Engineering), Air Force Institute of Technology (Dept. of Electrical Engineering), and the Naval Post Graduate School (Dept. of Electrical Engineering) also participate in the program.

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Electronic Systems Division (ESD), Air Force Avionics Laboratory (AFAL), Foreign Technology Division (FTD), Air Force Weapons Laboratory (AFWL), Armament Development and Test Center (ADTC), Air Force Communications Service (AFCS), Aerospace Defense Command (ADC), Hw USAF, Defense Communications Agency (DCA), Navy, Army, Aerospace Medical Division (AMD), and Federal Aviation Administration (FAA).

Further information about the RADC Post-Doctoral Program can be obtained from Jacob Scherer, RADC/RBC, Griffiss AFB, NY, 13441, telephone AV 487-2543, COMM (315) 330-2543.

DIGROUP DATA REDUCTION TECHNIQUES

1. INTRODUCTION AND SUMMARY OF RESULTS

The purpose of this study is to determine the best way to compress the output of two T1 PCM carrier terminals down into the bit stream normally required by one T1 terminal*. The equipment which performs this 2:1 compression is herein called a Digroup Data Reduction (DDR) system. The use of this system can save money because it doubles the number of speech channels that can be placed on a T1 digital transmission line and thereby lowers the per channel transmission cost. Even when fully loaded there is no noticeable degradation to the user.

A block diagram of the DDR system and how it is used is shown in Figure 1. Only one direction of transmission is shown. A complete DDR terminal consists of a DDR encoder and a DDR decoder. T1 carrier systems normally require transmission over one 1.544 mB line in each direction. By using the DDR terminal two complete T1 carrier systems require a single 1.544 mB line in each direction. As shown in the figure the DDR system can accommodate up to six dedicated data channels on which no compression is possible because the nature of the data is unrestricted. These data channels are obtained by designating up to six of the T1 speech channels as data ports. These data ports are designed to accept digital data. They are routed straight through the DDR system without compression. Switches on the DDR terminal designate which of the T1 speech channels are to be regarded as data channels and omitted for the data compression algorithms designed for speech.

*A T1 carrier system carries 24 two-way speech communication channels. It is widely used in military and civilian telephone networks.

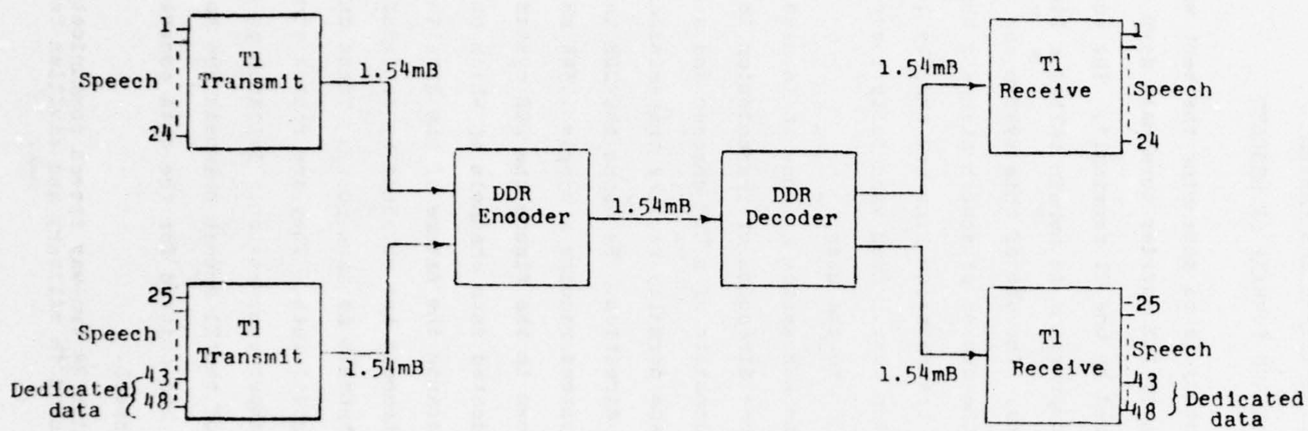

Figure 1. DDR system used for 2:1 compression of T1 PCM carrier systems

Chapter 2 of this report summarizes four previously proposed data compression systems that have been considered in the past for applications like this. Of these four only one seemed promising - the Speech Predictive Encoding Communication (SPEC) system currently being tested by COMSAT. In addition to SPEC we determined that two other promising possibilities existed - NIC/TASI, a marriage of Nearly Instantaneous Companding (NIC) with Time Assignment Speech Interpolation (TASI) and ADPCM/TASI, a marriage of Adaptive Differential Pulse Code Modulation (ADPCM) with TASI. These three systems - SPEC, NIC/TASI and ADPCM/TASI - were considered to be candidates for the DDR application and are given further study in Chapter IV. The last section of that chapter contains our reasons for choosing ADPCM/TASI as the system best suited for the DDR application.

The COMSAT SPEC system is reasonably well documented [1,2]. It is currently undergoing tests on satellite links. The concept of ADPCM is well known [3,4] but its use with TASI has not been previously explored in any detail. The TASI system itself has long been used on undersea analog cable transmission systems [5,6]. The NIC system is the newest speech data compression technique considered herein. It was presented at a conference in 1975 [7] as a possibility for mobile radio communications but the application to cable carrier systems was not evident until September 1976 [8] - only about two months before the termination of this study. All of the three candidate compression systems take advantage of two characteristics of speech: (1) the user of a two-way communication system talks only about 38% of time the system is off-hook (i.e., the duty factor is .38) and (2) speech sample values are slowly varying compared to the PCM sampling rate and are therefore somewhat predictable.

Since all three candidate systems take advantage of the average duty factor of speech, which is about .38, they must adapt to those times when the instantaneous duty factor is greater than .38; there are times when the system is heavily loaded and more than 38% of the users in one direction of transmission are talking. The SPEC system adapts to this by transmitting only the most unpredictable speech samples during high activity. Those samples which are frozen out are predicted by the decoder to be identical to the most recently transmitted value. The ADPCM/TASI and NIC/TASI systems adapt to high activity by reducing the number of bits used to encode each sample value. This technique is used in Europe on some PCM links.

The principle tool used in this study is computer simulation. Analyses of SPEC, ADPCM, TASI and, to a lesser degree NIC, are available in the published literature [1-8]. What we needed to determine was how these systems sound in a system designed for the DDR application. Computer simulation allowed us to programmatically vary a large number of the system parameters - such as system loading, duty factor, input signal level, number of dedicated data channels, etc. The simulations were done on an AGT/30 interactive computer. Telephone quality speech was sampled, coded and stored on a digital disk. Signal processing software which simulated various DDR systems was written (in Fortran IV) and the speech was passed through these programs. The processed speech could then be listened to and recorded on audio tape. No formal subjective tests were made but an audio demonstration tape was made and forms a part of this report. This audio tape is not furnished with every report but a copy

of the tape can be obtained if requested before November 31, 1977 - see Chapter 6 for ordering information and a description of the tape. The computer programs also furnished a number of objective performance measures such as signal-to-noise ratios. The computer simulations were used to compare three DDR systems:

(1) A lightly loaded system with only 16 off-hook users. The system performs best under this condition because the speech compression algorithms are not stressed. This system is called the 16/0 system.

(2) A fully loaded system with 46 off-hook telephone users and two dedicated data channels. This system is called the 46/2 system.

(3) A fully loaded system with 42 off-hook telephone users and six dedicated data channels. This is called the 42/6 system. The speech compression algorithms are highly stressed in this configuration. It represents a worst case condition - the speech compression is 2.3 to 1.

On the basis of these computer simulations and other considerations we feel that the ADPCM/TASI system is best for the DDR application. This conclusion reflects the fact that the ADPCM is better suited for compressing speech signals because it is more "tailored" to the statistics of speech than the other two candidate systems. The ADPCM system is not likely to operate well with non-speech signals such as voice band data. We have assumed that all data is presented to the data ports in digital form and that voice band data is not present on the speech channels.

2. REVIEW OF PREVIOUS DIGITAL SPEECH COMPRESSION SYSTEMS

Currently there is a lot of activity in the telecommunications industry in the design of systems for digital speech compression. Very complicated systems for this application can be constructed for reasonable cost using LSI.

Table 1 is a compilation of four systems that have been previously proposed for multichannel bit rate compression of digitally encoded speech. The last system in the table, SPEC, is being actively considered for applications in the telecommunications industry now and is a candidate for the DDR application. Following this table are brief outline descriptions of the systems listed in the table.

Table 1

Comparison of Previously Proposed Multichannel Digital Speech Compression Systems

System	Organization doing work	Compression Ratio	Effect of information overflow	Sensitivity to trans-mission errors	Complexity/ Cost	Has hard-ware been constructed & tested - date	Current Status of Technique
Vocoplexer ODVP (Run length coding)	Bell Aerospace	60% reduc-tion in bit rate for 48 channel system	graceful degradation PCM accuracy goes from 6 to 5 to 4 bits	High	Low	yes - 1972	inactive
PCM/TASI (Optimal digital voice trans-mission)	ITT	48 channel down to 20 (50% reduc-tion in B.R.)	Speech clipping & freeze out	Low-channel assignment protected against errors	Moderate	yes - 1973	inactive
Direct Entropy Coding of PCM levels	University of the Netherlands	slightly less than 2:1	graceful degradation similar to occasional freeze out	High	Moderate to low	No	unknown
SPEC	COMSAT	2:1	Freeze out over occurs in a sample time - graceful degradation	Moderate	Low development cost - much development work has been done	yes - 1974 and 1976	field tests in progress over satellite links

2.1 Vocoplexer or ODVP (Optimum Digital Voice Processor)

References: [12,13]

NOTES

1. Research sponsored by ECOM and RADC.

2. Digital data over troposcatter.

3. Constructed by Bell Aerospace.

4. Uses pauses in speech — 25% duty factor.

5. Codes output of 6 bit PCM to give 60% reduction — 48 speech channel PCM.

6. Run length coding starting with most sig bit of each channel. Probably very sensitive to errors — one error can cause errors in the rest of the frame (one bit error can cause catastrophic failure in all channels). Transmits sign bit as is.

7. Experiment was performed (field test).

8. Effect of buffer overflow is to reduce system to 5 bit PCM operation — graceful degradation.

9. Hardware model tested with 18 channel fully implemented plus simulator to simulate .25 activity factor. Also one 2.4 kB data channel.

10. Designed to operate without compander.

11. Effect of errors not considered in test model.

2.2 PCM/TASI

References: [10,11]

NOTES

1. 48 channel 6 bit PCM over TROPO (AN/MRC-98).
2. Compresses 48 channels down to 19 + 1 channel overhead.
3. Average length of freezeout is 32 m sec.
4. Each 125 μsec frame has 19 6 bit channel blocks + 8 bits for channel signaling which specifies channel assignment.
   1 multiframe = 40 frames.
   ∴ There are 40 x 8 = 320 bits for channel assignment.
   $\frac{320}{20}$ = 16 channel assignment bits per channel each multiframe.
   Use $\frac{6 \text{ bits}}{48 \text{ chan}}$ info + 10 bits redundancy.
5. BCH code with 7 info + 8 redundancy + 1 parity check bit for 7 info bits. Hamming distance of 6 ∴ 3 bit errors can be corrected. When error is detected channel is squelched — this means that channel assignment is not updated — previous channel assignment is used. This is like SPEC.
6. Channel assignment updated every 5 msec.

2.3 Direct Entropy Coding of PCM Levels

Reference : [9]

<u>NOTES</u>

1. Assign a simplified entropy code to PCM channels
   transmit 0 for inactive channels called a
   transmit 1 + 8 bit code for active channels 1-9 code
2. Achieve a rate of 4.4 bits per 8 bit sample.
3. Other entropy codes are possible — the 1-9 code is simple.
4. 16 bit framing and signaling word.
5. Like SPEC, they force a zero code in inactive channels.
6. Do not eliminate predictive redundancy from signal — use only activity factor.
7. No channel signaling information is considered.

2.4 SPEC

References: [1,2]

NOTES

1. Converts 64 speech channels down to 32 — 2.048 mB/sec.

2. Unlike other systems it takes some advantage of predictability of speech.

3. Digital voice switch used to eliminate noise during silent intervals. Only active speech goes through.

4. Uses zero order predictor.

5. Assignment recirculation unit rotates priority of service to input trunks.

6. 64 bit sample assignment word (SAW) contains a 1 for each active channel.

7. Sample is transmitted only if it exceeds previous sample by some δ — $\delta = 3$ is imperceptable from $\delta = 0$. $\delta = 3$ means 3 quantizing levels.

8. Service priority circulated over all trunks to give uniform degradation to all.

9. Single parity check on SAW.

10. Uses A law companding, $A = 87.6$.

11. Activity factor with $\delta = 0$ is 33% } 9% change.
    Activity factor with $\delta = 3$ is 24% }
    δ is called predictor aperture or threshold. δ changes from 0 (low loading) to 11 (high loading).

12. Busying out channels is not used — degradation is spread uniformly into all speech channels — there is no priority system. Data channels are not handled differently from speech.

13. Spec is equivalent to standard 8-bit PCM 95% of the time — field test (must vary with # and type of VB data sets on line).

14. Intelligible crosstalk cannot occur.

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3. THE TASI PRINCIPLE

TASI (Time Assignment Speech Interpolation) is a technique which allows c transmission channels to be shared among n speakers (or users). It takes advantage of the fact that a user of a speech communications channel talks, on the average, only about 38% of the time. For the remaining 62% of the time the user is either listening or pausing. Thus the user needs a transmission channel or path only 38% of the time. TASI takes advantage of this 38% duty factor by detecting when speech is present and connecting the speaker to a transmission channel only when he is speaking. Extensive studies have been made on talker statistics and the advantages to be gained using TASI [5,6].

The degradation usually associated with TASI systems is clipping on the initial sounds in a speech burst. The subjective effect of speech clipping is negligible if the clipping is less than about 50 m sec [14]. Thus the job of the TASI system is to determine when speech is present and switch the speaker to a transmission channel within 50 m sec. Some fraction of the bit rate in a digital TASI system must be devoted to transmitting this voice switch information from encoder to decoder so that the decoder knows which speaker is connected to each of the transmission channels. This voice switch information is contained in what we call the "channel assignment message".

In the DDR application one frame is 193 bits long and lasts for 125 μsec. At least once every 50 m sec (400 frames) the channel assignment message must inform the decoder of which of the 46 speech channels is on and should be assigned to a transmission channel. A simple 46 bit word can do this by transmitting a 1 in

each of the 46 positions if that voice switch is on. Otherwise a zero is transmitted. This requires only an average of $\frac{46}{400}$ = .11 bits per frame. In the systems described herein we devote two bits per frame for this voice switch information and two bits for error protection. Using a rate $\frac{1}{2}$ error correcting code for error protection, which is needed because errors in the assignment word are very disruptive, 4 bits per frame are used for the channel assignment message. This allows the voice switch information to be transmitted once every 23 frames (2.9 msec). In what follows this 4 bits per frame is allocated for the channel assignment message for both the 46 and 42 speech channel versions of the DDR system.

3.1 <u>Signaling over a Digital TASI System</u>

The PCM systems which are the inputs to the DDR compressor send one signaling bit every sixth frame or 1/6 bit per frame. For 46 input channels this is about 1/6 x 46=8 bits per frame. In the TASI systems simulated and studied herein we have assigned 8 bits of each 193-bit frame to be signaling bits. The signaling information is stripped off the 46 PCM channels and transferred to this 8-bit TDM signaling "channel".

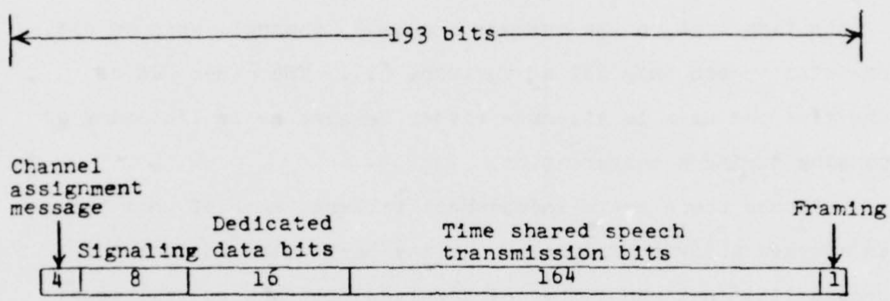
In this method the DDR system is transparent to the signaling information. In a more sophisticated DDR system we could take advantage of the fact that the PCM signaling information has a very low information rate. The maximum signaling rate for each speech channel is 20 pulses per sec. (actually 10pps is adequate for almost all systems). For a 20pps rate each dial pulse must be switched on and then switched off — this requires 40 bits/sec for 46 channels: 46 x 40=1840 bits/sec are needed. This is only about 1 pulse every 4 frames. This signaling information must be protected against transmission errors. Using a 1/2 rate error

correcting code would require 1 bit every two frames. This could be accomplished by using every other framing bit which in PCM systems is reserved for Common Channel Interoffice Signaling (CCIS). The T-carrier systems are built to frame on 1 framing bit every 2 frames. With this method the 8-bit per frame now required for signaling could be used for speech transmission.

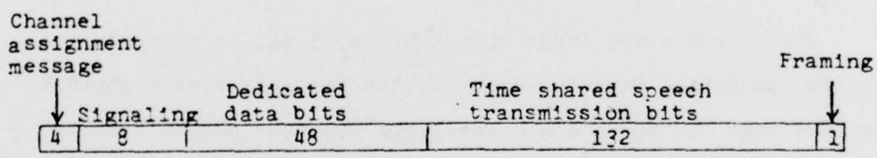
In an even more sophisticated arrangement both the signaling and voice switch information could be encoded into one 6-bit word per frame. This word could then be transmitted only when needed - i.e., only when voice switch or signaling information is required. In whatever signaling scheme is used the probability of a signaling error should be much lower than the p(e) provided on typical T1 lines. The telephone company uses error detection and retransmission for all of its CCIS channels. Error probabilities on the order of 10^{-10} are achieved on signaling channels of the toll plant of the Bell System. For the purpose of this study we have assigned 8 bits per frame to a separate signaling channel as discussed in the first paragraph of this section.

3.2 The Digital TASI Frame

In this study we have assumed that for each 125 μsec frame the voice switch information occupies a 4-bit channel assignment channel; the signaling occupies 8 bits; and the remaining 180 bits are devoted to data and speech transmission. The frame bit assignments for the 46/2 and 42/6 systems are shown in Figure 2.



(a) 46/2 TASI system



(b) 42/6 TASI system

Figure 2. TASI frame format

3.3 Talker Statistics

The purpose of this section is to present talker statistics for multiple channel carrier systems with 46 and 42 off-hook speech channels. The systems described herein take advantage of the fact that on the average a one-way channel, when in use, contains speech only 38% of the time [1]. The other 62% of the time the user is silent - either because he is listening or pausing during a conversation.

Assume there are n independent talkers, each of whom has an average activity factor p. At any particular instant the probability that the number of simultaneous talkers will equal or exceed c is given by the cumulative binomial distribution:

$$B(c,n,p) = \sum_{x=c}^{n} \frac{n!}{x!(n-x)!} p^x (1-p)^{n-x}$$

When c and n are large enough B(c,n,p) can be approximated by the cumulative Gaussian distribution and in the case where B is less than .50 and n and c are large B(c,n,p) can be computed by this formula [15]:

$$B(c,n,p) = \text{Exp}\left[-\frac{1}{\sqrt{2}}(1 + y + .6y^2)\right] \qquad (1)$$

where $y = \frac{c-np-1/2}{\sigma}$

$\sigma = \sqrt{np(1-p)}$

c = number of channels

p = average activity

n = number of talkers

This formula can be used to determine what percentage of the time c channels are active when n channels are off-hook.

For systems with n=46 and 42 off-hook channels the function B(c,n,p) is tabulated in Table 2 for some relevant values of c.

Table 2. Tabulation of B(c,n,.38)

c	n=46	n=42
14*	.67	
15*	.66	
16*	.65	
17	.59	.43
18	.49	.32
19	.380	.211
20	.272	.130
21	.180	.073
22	.110	.038
23	.063	.018
24	.033	.0079
25	.0159	.0032
26	.0071	.0012
27	.00294	.0004
34		1.8×10^{-8}

*These entries obtained from B(c,n,.38)=1-B(n-c,n,.62).

For systems with 46 and 42 off-hook channels the probability that exactly c channels are in use are contained in Table 3 for various values of c.

Table 3. Number of simultaneous talkers for system with 46 off-hook users.

c number of simultaneous talkers	Percentage of time c users are talking simultaneously	
	n=46 off-hook chan.	n=42 off-hook chan.
16 or less		57
17 or less	51	68
17	10	11
18	11	11
19	11	8.1
20	9.2	5.7
21	7.0	3.5
22	4.7	2.0
23	3.0	1.0
24	1.7	.46
25	.88	.20
26 or more	.71	.12

This table shows, for example, that, in a system with 46 off-hook speech channels, the probability is only .03 that

23 people will be talking simultaneously. The systems described herein take advantage of this.

Bits per sample on TASI-type systems

Using Table 3 we can calculate the probability that, say, L bits will be available to any given speech sample. Consideration of voice switch requirements has led to the conclusion that for the 46/2 system 164 bits are available per frame for the transmission of the speech samples. In this system 16 bits are being used for the dedicated data lines and 12 bits are used to transmit voice switch and signaling information. For the 42/6 system 132 bits are available for speech samples, 48 are used for data and 12 for voice switch and signaling information. The probability that L bits are available for speech samples is given in Table 4.

Table 4. Probability that L bits are available for each speech sample.

L = bits available for speech sample	46/2 system (164 bits available)	42/6 system (132 bits available)
8	.82	.57
7	.147	.22
6	.032	.19
5		.018
4 or less		.0004

Examination of Table 4 shows that for the 42/6 system 6 or more bits are available for each speech sample 98% of the time. For the 46/6 system 6 or more bits are available 99.9% of the time.

3.4 High Activity Factor Statistics

Although the systems in this report are compared assuming a 38% activity factor, there are times when the activity factor exceeds this value for minutes at a time. In COMSAT's report

on the field trial of their SPEC system it is shown that during one three minute period the activity factor exceeded 45% (see Fig. 4(b) of [2]). There were several other shorter periods when the activity factor was about 45%. In mililary situations it seems likely that in some circumstances the activity factor in one direction or another might exceed 45%. For example, if there were significant military activity in one geographical region then telephone circuits coming out of this region would tend to have a higher activity factor than those circuits directed into the region; i.e., if this geographical region is generating military information then users in the region may tend to be more active talkers, whereas the people they are conversing with would tend to be listeners.

Although this issue is not considered further in this report talker statistics have been computed for an activity factor of 45% for the 46 and 42 channel systems considered herein. Table 5 shows the quantity B(c,n,.45) for n=46 and n=42. Table 6 shows the percentage of time c users are talking simultaneously for these two systems.

Table 5. Tabulation of B(c,n,.45)

c	n=46	n=42
19		.535
20		.426
21	.513	.312
22	.407	.211
23	.300	.1319
24	.205	.0758
25	.1298	.0401
26	.0764	.01959
27	.0417	.00881
28	.02117	.00365
29	.00996	.001395
30	.00435	.000492
31	.001765	
32	.000664	

c=number of simultaneous talkers	Percentage of time c users are talking simultaneously n=46 off-hook chan.	n=42 off-hook chan.
18 or less		.465
19		.109
20 or less	.487	.688
20	.096	.114
21	.106	.101
22	.107	.079
23	.095	.056
24	.0752	.0357
25	.0534	.0205
26	.0347	.0108
27	.0205	.00516
28	.0112	.00225
29	.00561	.0009
30	.00258	
31	.00110	

Table 6. User statistics for the high activity factor of .45.

4. CANDIDATE DDR SYSTEMS

In our judgement there are three candidate systems which could best satisfy the task of combining two T1 24 channel terminals into the 1.54mb bit stream normally occupied by one terminal. These are

1. SPEC (Speech Predictive Encoding Communication system) proposed by COMSAT
2. NIC/TASI (Nearly Instantaneous Companding, a form of adaptive pulse code modulation combined with Time Assignment Speech Interpolation)
3. ADPCM/TASI (Adaptive Differential Pulse Code Modulation combined with TASI)

At the time of this writing COMSAT has published a considerable amount of information [1,2] about earlier versions of SPEC although recent improvements are not documented. Nothing has been published about the NIC/TASI system by Bell Telephone Laboratories which is apparently performing tests on such a system between New York and Boston [16]. Two articles [7,8] discuss NIC but not in conjunction with TASI. There is considerable information on ADPCM [3,4] but none on its use with TASI. Operating with only speech signals, computer simulation has shown that ADPCM/TASI can achieve speech compression ratios of up to 3:1 before any degradation is perceived by a normal user.

4.1 SPEC

The SPEC (Speech Predictive Encoding Communication) system was conceived in 1972 by COMSAT to improve the efficiency of

satellite links. It was originally designed to accept a 4.096 Mb/s TDM frame consisting of 64 PMC voice samples every 125 μsec, a European standard. The output rate of the system is 2.048 Mb/s yielding a bit rate compression ratio of 2:1. Subsequent prototype models of the system could achieve a 2:1 compression with any number of input PCM voice channels, e.g., it could compress 48 channels down into the bit rate of 24. The SPEC system is described in [1] and field test results over INTELSAT satellite links are contained in [2]. Other field tests are currently in progress. In what follows we summarize the configuration of the SPEC system when used to compress the output of two T1 terminals down into a 1.544 Mb/s digital signal normally required by one T1 terminal.

A diagram of the SPEC transmitter when used in the DDR application is shown in Fig. 3. The two T1 carrier systems shown contain 48 voice band channels. Two of these channels are permanently assigned to data and are not affected by the SPEC system*. A time-shared digital voice switch continuously determines which of the 46 input voice channels contain active speech in order to prevent unnecessary transmission of noise during silent intervals. The zero-order predictor examines each sample from each input trunk and sends only unpredictable PCM samples along with a sample assignment word to the output multiplexer. A simple parity check is generated by the error control unit to protect the assignment word. The assignment recirculation unit rotates the priority of service to the

*Actually for the DDR application from 2 to 6 channels are dedicated to data. This explanation assumes only two data channels, however.

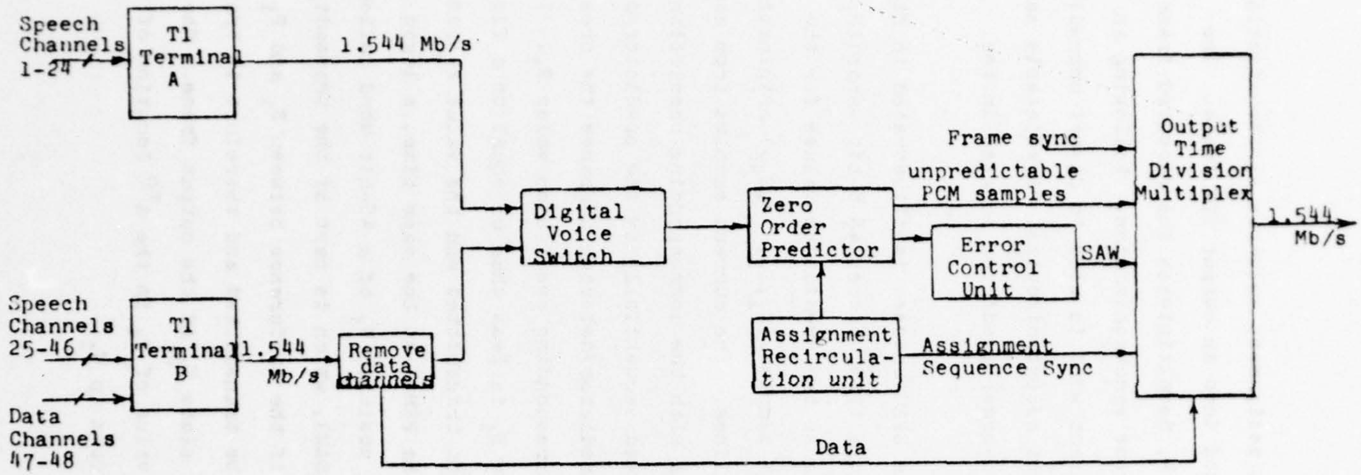


Figure 3. SPEC transmitter

input trunks. Frame and assignment sequence synchronization codes are also multiplexed into an output TDM frame. The receiver, shown in Fig. 4, demultiplexes the received frame and performs frame and sequence synchronization. Following an error check, the sample assignment word is used to direct unpredictable PCM samples to the correct output circuits. Predictable samples are regenerated by a zero-order predictor located in the reconstruction decoder.

The operation of the SPEC encoder is illustrated in Fig. 5. The predictive frame memory (PFM) stores 46 8-bit words ($P_1, P_2, \cdots, P_{46}$), which are used as the prediction values for the corresponding current PCM samples ($S_1, S_2, \cdots, S_{46}$) originating from the two T1 carrier lines. The current samples from each trunk $S_1, S_2, \cdots, S_{46}$ along with the corresponding prediction values $P_1, P_2, \cdots, P_{46}$ are processed sequentially by the predictor/detector. For the i^{th} trunk, the predictor/detector compares the present PCM samples S_i to its corresponding prediction value P_i. If the difference between S_i and P_i is less than or equal to a fixed threshold δ then S_i is not transmitted and the value P_i remains in the i^{th} location of the PFM. At the same time, a logic 0 is written into the i^{th} bit position W_i of a 46-bit word called the sample assignment word (SAW), which is part of the transmitted TDM frame. Conversely, if the difference between S_i and P_i exceeds δ, then S_i must be transmitted and therefore is transferred to one of the 8-bit time slots T_j of the output frame. The value of S_i also replaces the value of P_i in the i^{th} location of the PFM and a logic 1 is written in W_i.

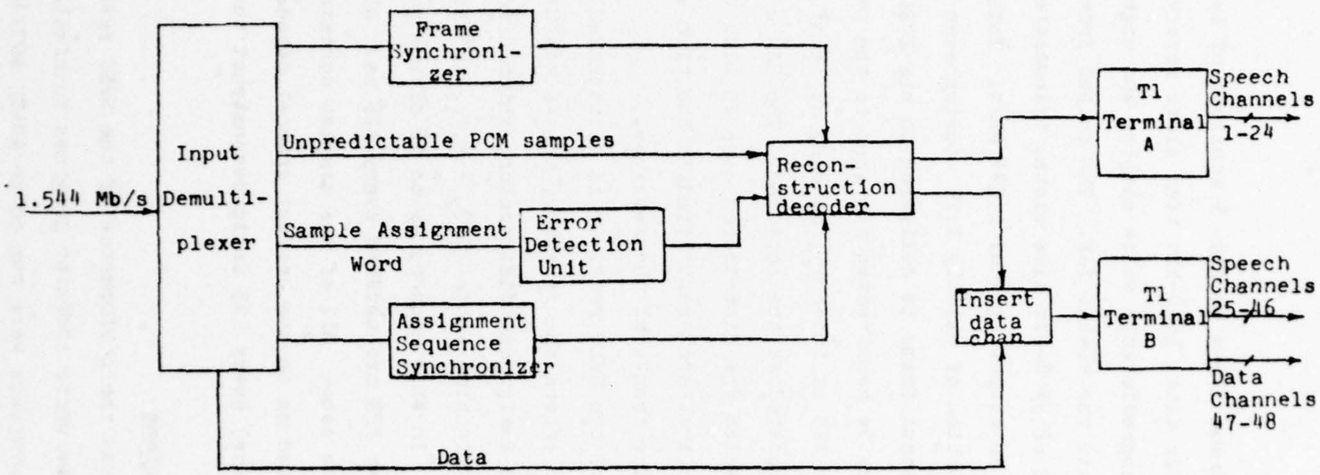


Figure 4. SPEC receiver

The output TDM frame shown in Fig. 5 consists of two 8-bit time slots reserved for data, 16 8-bit time slots reserved for transmission of the unpredictable voice samples and eight 8-bit time slots that contain the 46-bit SAW. The output frame is therefore composed of 24 8-bit time slots, transmitted in 125 μs, resulting in an output rate of 1.544 Mb/s. This represents a bit-rate reduction of exactly 2:1. During each 125 μs interval, a new TDM input frame is delivered to the SPEC system and a TDM output frame is constructed and sent to the receiver. Therefore, during any 125 μs interval, as many as 16 of the 46 voice trunks can be updated at the receiver. The SAW contains up to 16 logic 1's, which are distributed over 46 bits to direct the unpredictable samples that are transmitted in the time slots T_j to the correct outgoing trunks at the receiver.

This operation of the SPEC receiver is illustrated in Fig. 6. As the transmit frame enters the receiver, the 46 bit SAW is delivered to a sample assignment distributor, which directs the new samples contained in time slots $T_1, T_2, \cdots, T_{16}$ to the correct location in the PFM. In this manner, up to 16 of the samples stored in the receiver PFM are updated every 125 μs. All samples not updated remain the same. All of the samples contained in the PFM are then passed on to the correct channel decoders (in the $1, 2, \cdots, 46$ sequence) every 125 μs to reconstruct the speech signals.

SPEC Computer Simulations

In order to access the performance of the SPEC system in the DDR application we wrote computer programs to simulate its performance. These programs were run on an ADAGE AGT/30 computer and objective measurements made of SPEC performance. Audio tapes

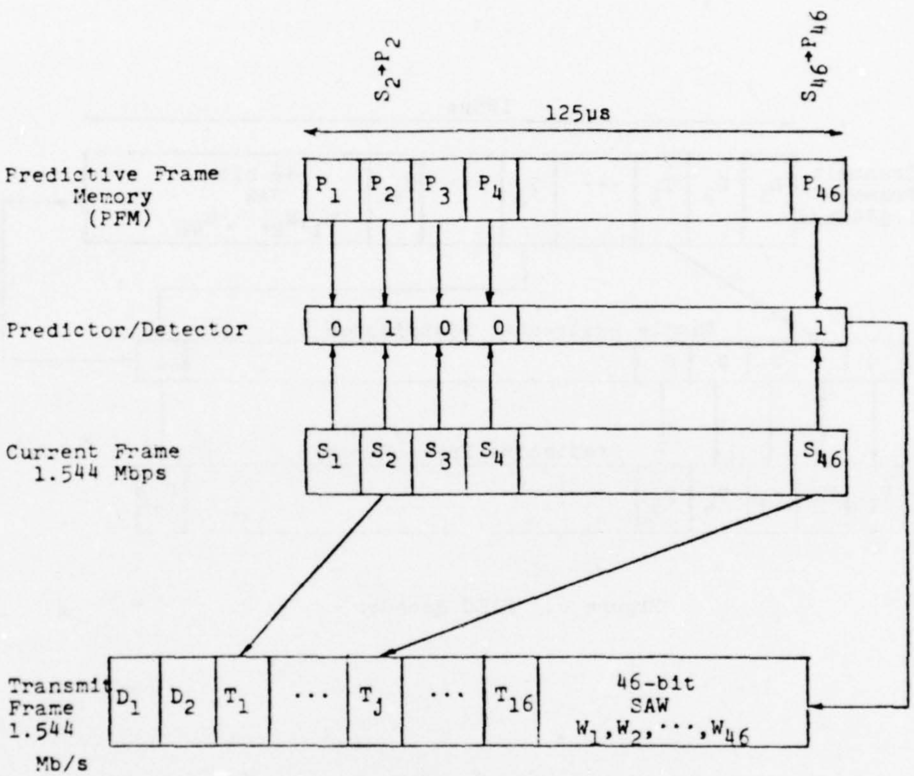


Figure 5. SPEC encoder and frame

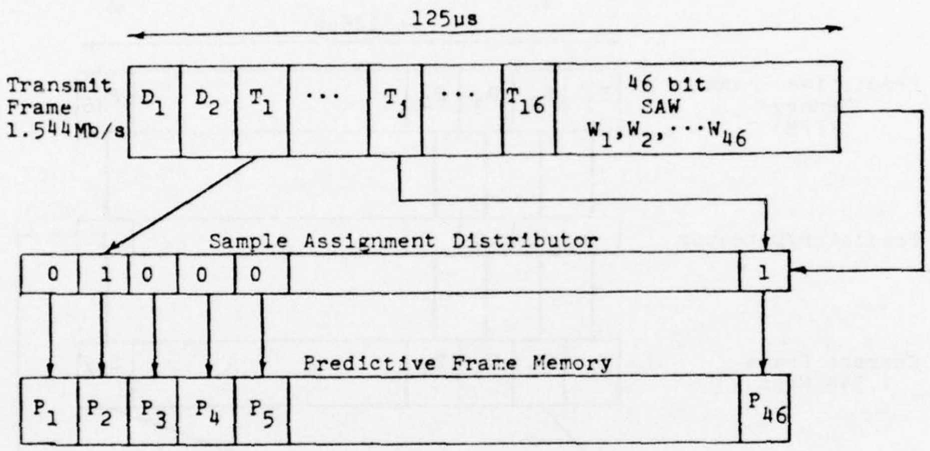


Figure 6. SPEC decoder

were also made to demonstrate system performance. These tapes form a part of this report. The program used is called SPEC3. A listing of this program is contained in Appendix A of this report.

We simulated three SPEC systems for the DDR application. Their properties are summarized in Table 7 below.

SPEC System	Active Speech Channels	Permanent Data Channels	Time shared 8-bit speech slots/frame
46/2	46	2	16
42/6	42	6	12
16/0	16	0	16

Table 7. SPEC Systems Simulated

The 46/2 system has 46 active (off-hook) speech channels and two data channels. The 46 speech channels share 16 8-bit transmission slots. Similarly the 42/2 system has 42 active speech channels, six data channels and 12 8-bit transmission slots. The 16/0 system has only 16 active talkers all of which are assigned to an 8-bit transmission slot. The 16/0 system shows the maximum performance of SPEC during low traffic conditions. Its performance is identical to that of 8-bit μ=255 PCM. We did not consider signaling, supervision or alarm arrangements for these systems. The 46/2 system is the one described in the previous section. The SPEC 3 program simulated systems which agreed in every respect except two with the system described in [1]. These two exceptions are outlined in Appendix B. Both changes improved the performance of the system.

The SPEC 3 program contains subroutines which simulate traffic loading statistics.

The following eleven speech segments were used as source material for the simulations:

1. Nine rows of soldiers stood in line. (male #1)
2. The beach is dry and shallow at low tide. (male #1)
3. The pipe began to rust while new. (female speaker)
4. Open the crate but don't break the glass. (female speaker)
5. Act on these orders with great speed. (female speaker)
6. Add the sum to the product of these three. (male #2)
7. Thieves who rob friends deserve jail. (male #2)
8. The hog crawled under the high fence. (male #2)
9. Cats and dogs each hate the other. (male #3)
10. Open the crate but don't break the glass. (male #3)
11. Thieves who rob friends deserve jail. (male #3)

Segments 1 and 2 were made using a high quality microphone. Segments 3-11 are copies of sentences recorded at Bell Telephone Laboratories through a 500 type telephone set. These sentences were filtered to 3400 Hz and sampled at 8 KHz. Each sample value was converted into a 15-bit number using a uniform quantizer. These 15-bit words were stored on disk in the AGT/30 computer and used as source material for the encoding systems simulated on this contract. Table 8 shows some of the details of these sentences when run through an 8-bit μ=255 companded PCM system. Since the sampling rate is 8 KHz the number of samples shows that each segment is from two to three seconds long. The signal-to-quantizing-noise ratios of about 37dB verify that the 8-bit μ=255 PCM system is ideal. The S/N* ratios are almost identical for all speech segments. The signal power merely reflects the

*In this report the term S/N refers to the ratio of signal power to unweighted noise power. In general "C" message weighting of the noise adds 2.9dB to this ratio.

Table 8. Source material for computer simulations

Speech Segment	No. of Samples	8-bit PCM S/N in dB	Sig Power in dBm	Power relative to a full load sinusoid
1	23219	37.2	-16.8	-13.8dB
2	23057	37.4	-16.3	-13.3dB
3	15765	37.4	-16.9	-13.9dB
4	19025	37.2	-15.5	-12.5dB
5	17609	37.2	-17.1	-14.1dB
6	21465	37.3	-19.7	-16.7dB
7	21151	37.3	-19.7	-16.7dB
8	20629	37.5	-18.3	-15.3dB
9	16779	37.2	-20.8	-17.8dB
10	16931	37.5	-20.0	-17.0dB
11	17361	37.2	-19.5	-16.5dB

computer scaling of the sample values. The column entitled "Power relative to a full load sinusoid" shows how the stored speech segment is matched to the simulated PCM system. The power in the signal can be changed programmatically before passing the signals through any of the computer simulations.

Figure 7 shows how the performance of the three SPEC systems varies with input power level. The curves show the dynamic range of about 35dB typical of μ=255 PCM systems. The 16/0 system is operating like an ideal PCM system while the 46/2 and 42/6 systems have maximum S/N ratios of about 29 and 23dB respectively.

Figure 8 shows how the performance of the SPEC systems changes with the number of off-hook channels. The 46/2 system has 16 8-bit transmission slots per frame while the 42/6 system has only 12. This is the only figure in which the systems are not fully loaded. The fact that these computer simulations don't lie on a smooth curve and seem slightly erratic is due to the fact that each point represents a simulation of only 5 seconds of speech.

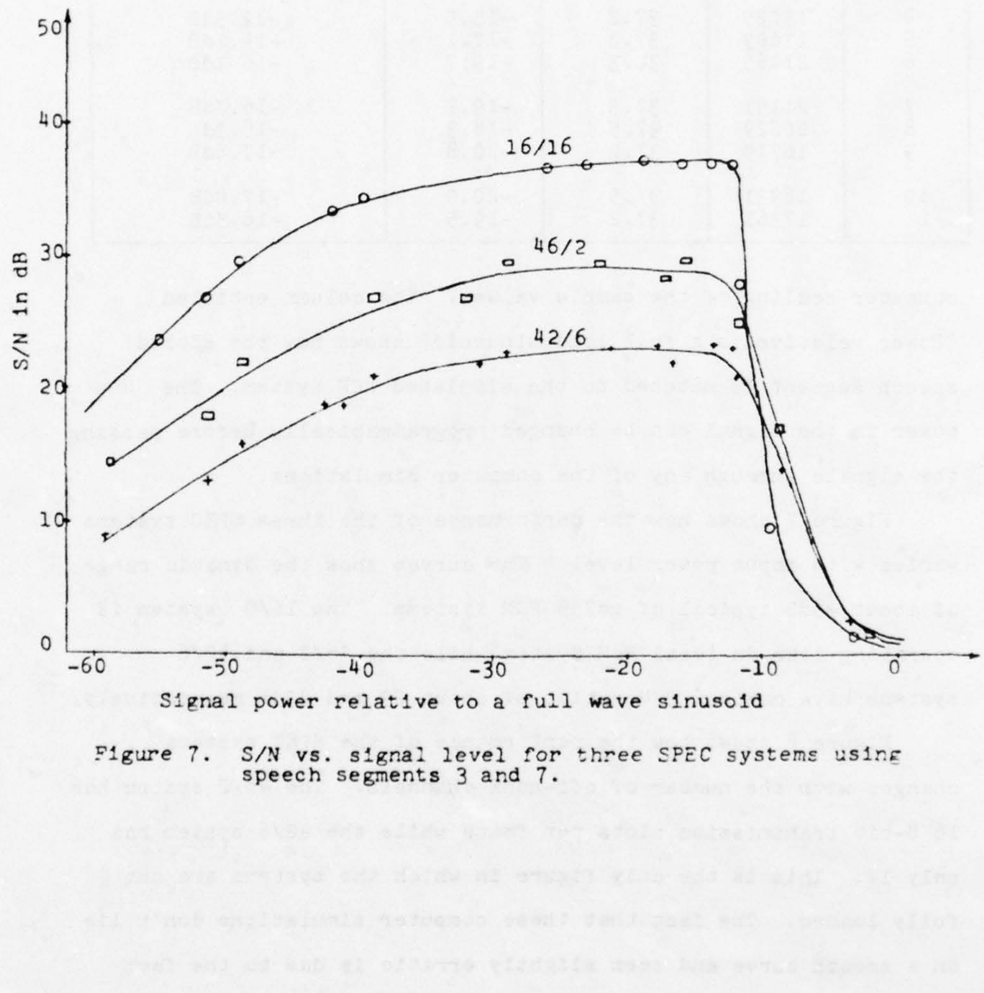


Figure 7. S/N vs. signal level for three SPEC systems using speech segments 3 and 7.

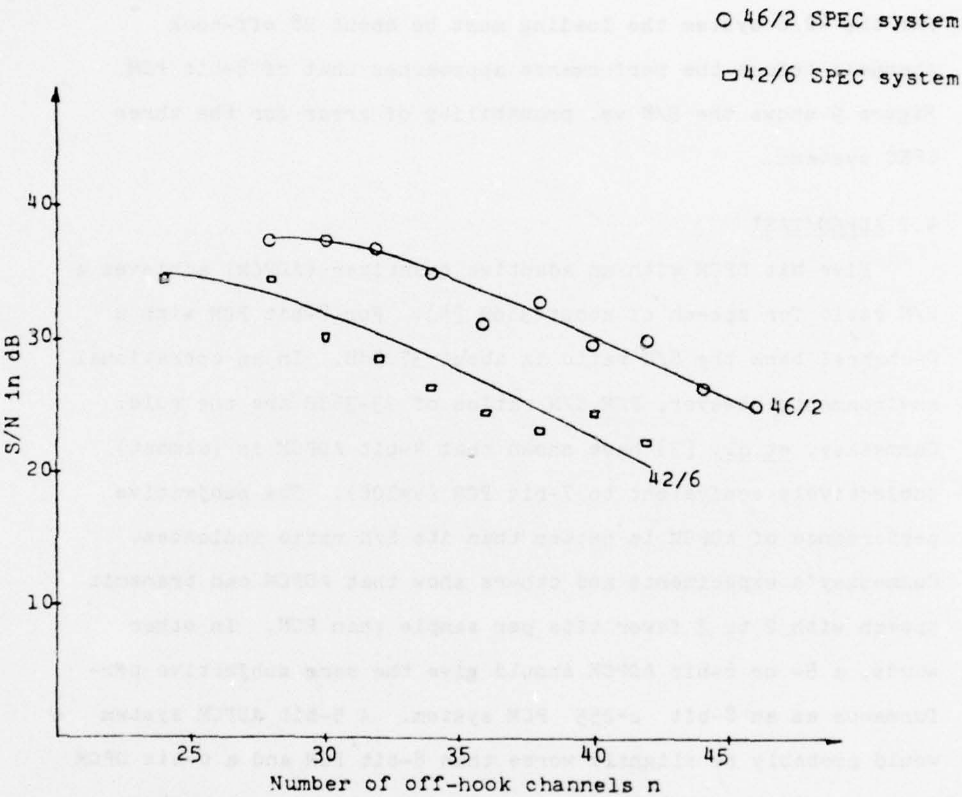


Figure 8. S/N ratio vs. number of off-hook channels for the 46/2 and 42/6 SPEC systems. Input is speech segments 3 and 7.

For the 46/2 system once the loading goes down to about 34 off-hook channels the system performance approaches that of 8-bit PCM. For the 42/6 system the loading must be about 28 off-hook channels before the performance approaches that of 8-bit PCM. Figure 9 shows the S/N vs. probability of error for the three SPEC systems.

4.2 ADPCM/TASI

Five bit DPCM with an adaptive quantizer (ADPCM) achieves a S/N ratio for speech of about 31dB [4]. For 8-bit PCM with a D-channel bank the S/N ratio is about 37.5dB. In an operational environment, however, PCM S/N ratios of 33-36dB are the rule. Cummeskey, et.al. [3] have shown that 4-bit ADPCM is (almost) subjectively equivalent to 7-bit PCM (μ=100). The subjective performance of ADPCM is better than its S/N ratio indicates. Cummeskey's experiments and others show that ADPCM can transmit speech with 2 to 3 fewer bits per sample than PCM. In other words, a 5- or 6-bit ADPCM should give the same subjective performance as an 8-bit μ=255 PCM system. A 5-bit ADPCM system would probably be slightly worse than 8-bit PCM and a 6-bit DPCM system would be slightly better. These conclusions have been verified by computer simulations run here at NCSU using the computer program JADPCM.

When used with the TASI system described in Chapter III of this report it is necessary for the ADPCM system to be able to drop bits during periods of high activity. Thus the ADPCM/TASI system may have word lengths of 8, 7, 6, 5 or even fewer bits depending on the number of active talkers. As with SPEC a voice switch determines whether or not speech is present on each

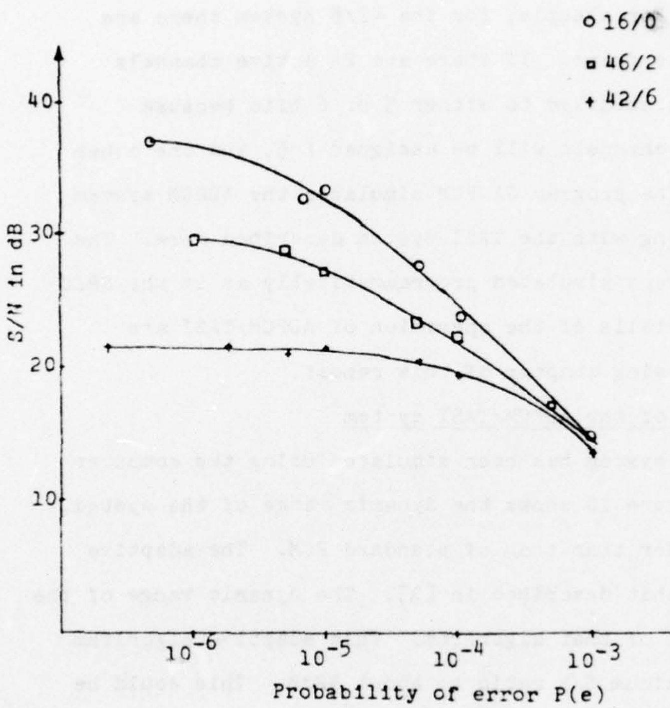


Figure 9. S/N ratio vs. P(e) for various SPEC systems using speech segments 3 and 7.

input channel. If it is, the ADPCM output is connected to a transmission channel. If not, no sample is transmitted. For each frame the system must determine how many bits are available for each sample value. For example, for the 42/6 system there are 132 bits per 125 μsec frame. If there are 24 active channels then each channel is entitled to either 5 or 6 bits because $\frac{132}{24} = 5\frac{1}{2}$. Thus 12 channels will be assigned L=5, and the other 12 will have L=6. The program JADPCM simulates the ADPCM system described in [3] along with the TASI system described here. The loading statistics were simulated programmatically as in the SPEC3 program. Further details of the operation of ADPCM/TASI are covered in the following chapter of this report.

Computer simulation of the ADPCM/TASI system

The ADPCM/TASI system has been simulated using the computer program MCDAQ2. Figure 10 shows the dynamic range of the system. It is somewhat broader than that of standard PCM. The adaptive algorithm used was that described in [3]. The dynamic range of the system is a function of that algorithm. This adaptive algorithm also limited the maximum S/N ratio to about 32dB. This could be increased to 37dB.

Figure 11 shows the performance of the system vs. number of off-hook channels for speech segments 3 and 10. The 16/0 system has the same S/N ratio as the 46/2 system and is not plotted.

Figure 12 shows how the expected performance of the system varies as a function of the probability of error on the line. The MCDAQ2 program does not, at this time, have the capability of introducing errors. We do have a program JADPCM which has the same ADPCM algorithm without the variable step size capability and JADPCM can simulate errors on the line. In Fig. 12 we show the performance of ADPCM with 4 and 5 bits per sample value.

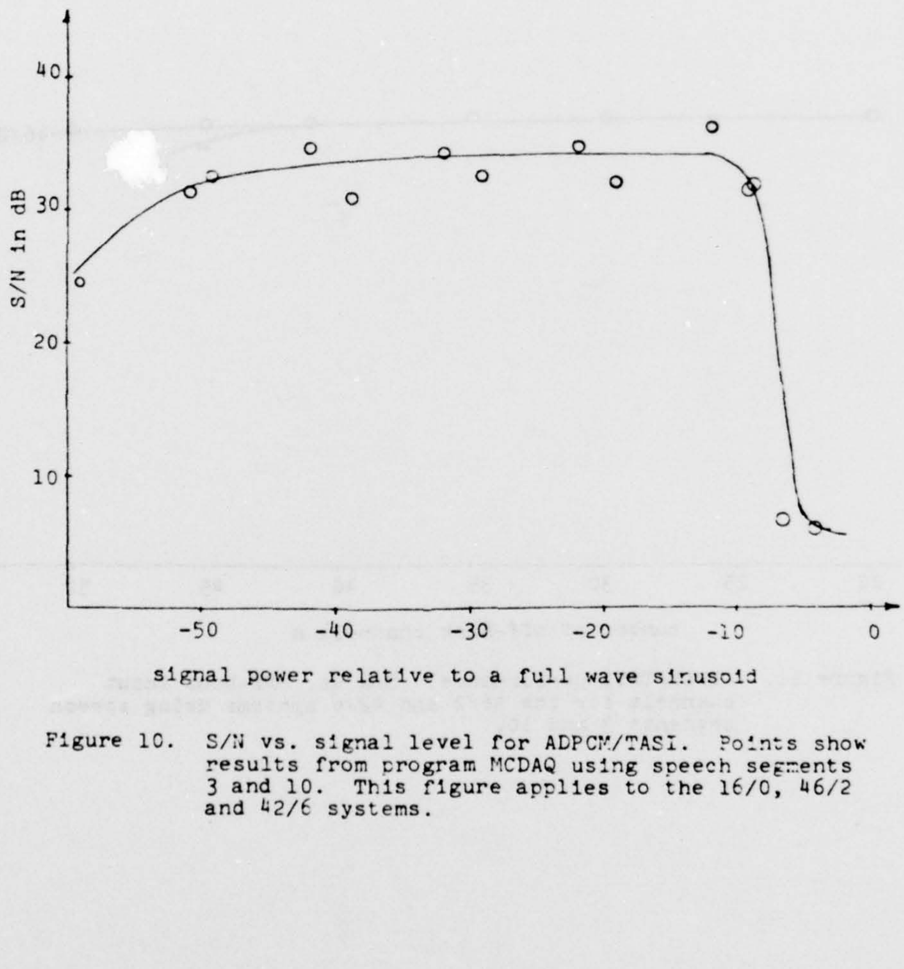


Figure 10. S/N vs. signal level for ADPCM/TASI. Points show results from program MCDAQ using speech segments 3 and 10. This figure applies to the 16/0, 46/2 and 42/6 systems.

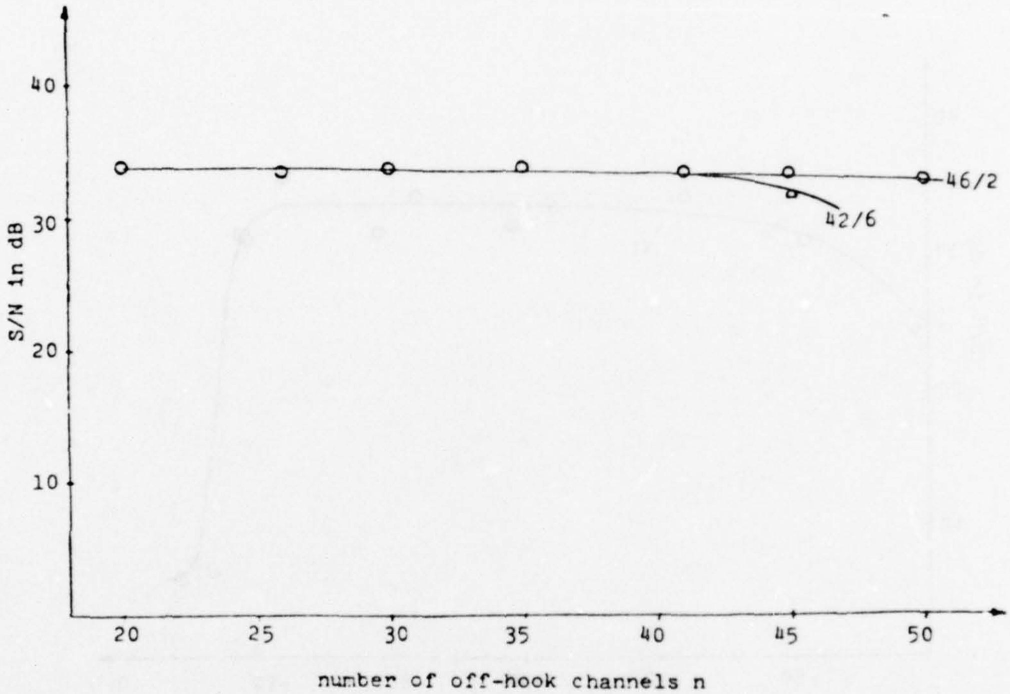


Figure 11. ADPCM/TASI performance. S/N vs. off-hook input channels for the 46/2 and 42/6 systems using speech segments 3 and 10.

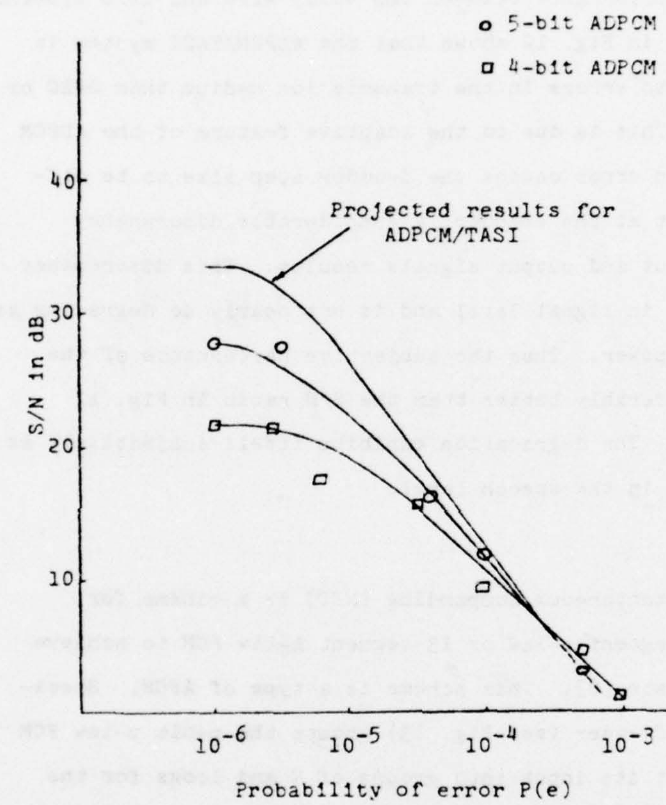


Figure 12. Probability of error vs. S/N ratio for ADPCM. Points show results from computer simulation using program JADPCM. Also shown are results projected for the ADPCM/TASI system. The 46/2, 42/6 and 16/0 systems would have identical performance.

From these results we have extrapolated the expected error performance of the ADPCM/TASI system. There is no essential difference in performance between the 46/2, 42/6 and 16/0 systems. The S/N plotted in Fig. 12 shows that the ADPCM/TASI system is more sensitive to errors in the transmission medium than SPEC or standard PCM. This is due to the adaptive feature of the ADPCM system. When an error causes the decoder step size to be different from that at the encoder, a considerable discrepancy between the input and output signals results. This discrepancy is a difference in signal level and is not nearly so degrading as noise of equal power. Thus the subjective performance of the system is considerably better than the S/N ratio in Fig. 12 would indicate. The degradation exhibits itself subjectively as a random change in the speech level.

4.3 NIC/TASI

Nearly Instantaneous Companding (NIC) is a scheme for processing 15-segment μ-law or 13-segment A-law PCM to achieve a reduced bit rate [8]. This scheme is a type of APCM. Specifically, the NIC coder (see Fig. 13) groups the n-bit μ-law PCM encoded words at its input into groups of N and looks for the sample having the largest magnitude in the group. It then re-encodes each sample in the group to approximately n-2 bit uniform quantization with overload at the top of the chord of the maximum magnitude sample. These n-2 bit uniform words are transmitted to a complimentary NIC decoder that regenerates n-bit codes in the original companding law. Since three scaling bits giving the chord of the maximum-magnitude sample must also be transmitted

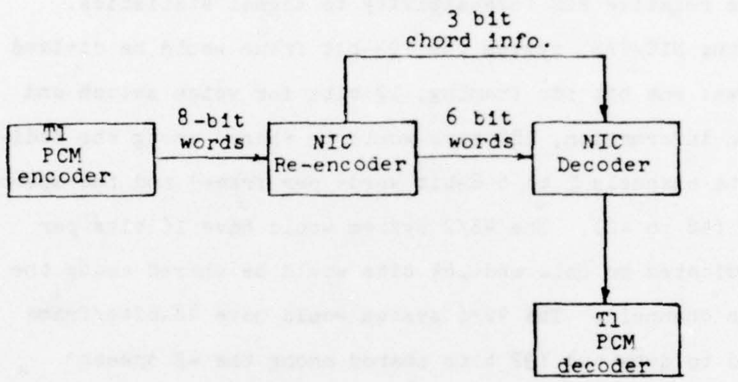


Figure 13. Processing 8-bit PCM with the NIC algorithm.

to the decoder, the resulting bit rate is $f_s(n-2+3/N)$ where f_s is the sampling rate of 8 KHz. This bit rate is comparable to that of n-2 bit PCM for even modestly large N. For the DDR system we use N=10. The dynamic range of the original PCM companding law is retained through the adaptive overload point as is the relative PCM insensitivity to signal statistics.

In the NIC/TASI system the 193-bit frame would be divided as follows: one bit for framing, 12 bits for voice switch and signaling information, 180 bits would be shared among the dedicated data channels (2 to 6 8-bit words per frame) and the speech channels (46 to 42). The 46/2 system would have 16 bits per frame dedicated to data and 164 bits would be shared among the 46 speech channels. The 42/6 system would have 48 bits/frame dedicated to data and 132 bits shared among the 42 speech channels. This is the same framing arrangement shown in Fig. 2.

For an NIC system with N=10 the number of bits per speech sample is, on the average, $6+\frac{3}{10} = 6.3$. For the 46/2 system the number of bits available per frame for speech samples is 164. If 26 channels are active the number of bits required is $26 \times 6.3 = 163.8$ — thus if 26 or fewer channels are active the NIC/TASI system will operate as an NIC system. If more than 26 channels are active then some bits must be dropped from the 6-bit NIC word. The probability of this happening is the probability that 27 or more channels are active at the same time. The probability of this happening is $B(27,46,.38) = .003$. Thus for the 46/2 system a bit would seldom be dropped and the NIC/TASI system should perform like standard NIC. For the 42/6 system there are 132 bits per frame and, since $6.3 \times 20 = 126$, 21 channels would have to be active before bit dropping is necessary.

The probability that a bit would need to be dropped on a given speech sample is B(21,42,.38) = .038. Since this is such a small percentage of the time it would have little effect on the S/N ratio and again the performance of the system would be very nearly like that of NIC without companding.

Unfortunately there is very little published information about NIC. We became aware that NIC is a possible candidate system for the DDR in September when the August 1976 issue of the IEEE Transactions on Communcations became available [8]. No subjective tests on the system have been published and the performance of the system in the presence of errors is not documented. Thus at the present time [11/29/76] we know only that it may be a promising system for bit rate reduction.*

4.4 Comparison of Candidate Systems

In this section we discuss the relative merits of the three candidate systems, SPEC, NIC/TASI and ADPCM/TASI and describe the reasons for choosing the ADPCM/TASI system in preference to the other two.

We have compared these systems based on the five criteria listed in Table 9. In each column of the table the performance of each system is rated relative to the other two — a rating of 1 means most desirable and 3 is least desirable.

Table 9. Comparison of candidate DDR systems

Attribute	ADPCM/TASI	NIC/TASI	SPEC
Equipment complexity	2	1	1
Performance with light traffic load	1	1	1
Performance with heavy traffic load	1	3	2
Performance in presence of errors	2	1	1
Possibilities for further compression	1	3	2

*Since the text of this report was written we have simulated the NIC system and found its performance to be less than that of SPEC for the DDR application.

On the basis of equipment complexity the ADPCM/TASI system is more complex than the other two and is rated 2 while SPEC and NIC/TASI seem to be equally complex and are rated 1. Each system requires essentially the same voice switch hardware and this will be a major cost of the system. The ADPCM system must contain hardware for the differential operation, which is difficult with a companded input and step size adaptation. This is the primary reason it is more complex than the other two systems. The NIC/TASI system must contain a procedure for selecting the largest sample in a group of ten and some sort of buffering and multiplexing arrangement to transmit the 3-bit chord information every 10 words. The SPEC system must contain a priority selection procedure and hardware for the variable threshold arrangement. All three systems must extract signaling from the PCM words every sixth frame and reencode these bits into a signaling channel which must be multiplexed into the bit stream. All three systems must derive, multiplex and transmit channel assignment information. In addition to this the NIC/TASI and ADPCM/TASI systems must protect their channel assignment information with an error correcting code. The alarm, framing and interfacing hardware are alike for all three systems. If such systems were ultimately realized through LSI, the cost of all three systems would be essentially the same even if there are differences in their complexity.

The performance of the systems during heavy load could be quite different. Our computer simulations show that ADPCM/TASI has no perceivable degradation even with all channels off-hook for both the 46/2 and the 42/6 versions of the DDR. The SPEC system simulations show that the noise is noticeable with the

46/2 system and objectionable with the 42/6 system. Objective S/N comparisons shown in Fig. 7 for SPEC and Fig. 10 for ADPCM/TASI verify the performance advantage of ADPCM/TASI. The NIC/TASI system has recently been simulated here at NCSU and at Bell Telephone Laboratories [8]. Their conclusion is that the objective S/N ratio of NIC is about 3dB worse than could be provided by ADPCM. Other simulations of ADPCM [4] which have been verified here show that ADPCM is subjectively about 3dB better than its objectively measured S/N ratio indicates. In comparison to PCM for example an ADPCM system with a S/N ratio about 3dB below that of a PCM system will be subjectively equivalent to the PCM system. Extrapolation from these subjective observations shows that ADPCM/TASI will sound 6dB better than NIC/TASI even though its S/N ratio may only be 3dB higher. This is significant for a heavily loaded system. For this reason we rated ADPCM/TASI "1", NIC/TASI "3" (although little is known about it) and SPEC "2".

The performance of the systems during a light load are subjectively equivalent - noise will not be perceived on any of them.

On performance in the presence of errors we rated SPEC "1", ADPCM/TASI "2" and NIC/TASI "1". Figs. 9 and 12 show that the objectively measured S/N ratio for ADPCM/TASI is slightly worse than the performance of SPEC. The error performance curves for NIC have not been published. It has been surmised, however, that NIC has an error performance like that of PCM [8]. If this is true then NIC/TASI may have better performance in the presence of errors than either of the other two competing systems.

On possibilities for further compression ADPCM/TASI is clearly better than the other two systems. Computer simulations of this system were made for a compression ratio of 3:1 and 4:1. The 3:1 compression system compressed three T1 systems down into the bit stream normally occupied by one system. No degradation is perceived. The S/N ratio is greater than 30dB. Simulations were also run for 4:1 compression — some noise can be detected but it is not objectionable. Subjective comparisons with SPEC show that the ADPCM/TASI is clearly superior. The NIC/TASI system was rated "3" in this category. Even though very little is known about this system we feel it would perform better than SPEC because SPEC would tend to freeze out with very heavy loading while NIC/TASI will always transmit some bits for each channel during every sampling interval.

Making a choice between these three candidate systems is not easy. They each have favorable aspects. However, because of its good performance under heavy load and its potential for even further compression we feel that the ADPCM/TASI system is the best choice for the DDR application.

5. PRELIMINARY DESIGN OF AN ADPCM/TASI SYSTEM

5.1 Block Diagrams

Figures 1, 2 and 3 illustrate a preliminary design of an ADPCM/TASI system for the DDR application. For the purposes of this discussion we assume that the system is designed to transmit up to six 64 K bit/sec data channels.

Figure 2 shows the frame transmitted by the ADPCM/TASI system. As in PCM each frame consists of 193 bits which occur every 125μsec. This 125μsec is the inverse of the PCM sampling rate of 8 KHz. One bit every frame is devoted to framing. The remaining 192 bits are divided into three separate channels — a signaling channel of 8 bits, a channel assignment signal of 2 bits, data channels consisting of six 8-bit words for a total of 48 bits and speech transmission bits consisting of the remaining 132 bits. For the purposes of this discussion we have assumed that there are six 8-bit data channels per frame as shown in Fig. 1(b). Actually this is adjustable from 0 to six 8-bit words per frame. The signaling channel and channel assignment signals always contain 8 and 4 bits per frame, respectively. The number of speech transmission bits can be as high as 180 when there are no dedicated data channels or as low as 132 when there are six data channels.

The encoder of Fig. 14 shows that the two T1 inputs have a common clock. If this common clock is not provided then the T1 input lines must be buffered at the input of the encoder and bit stuffing techniques must be used to insure that the 1.544 mB lines are in sync. This can be done but hopefully is not necessary.

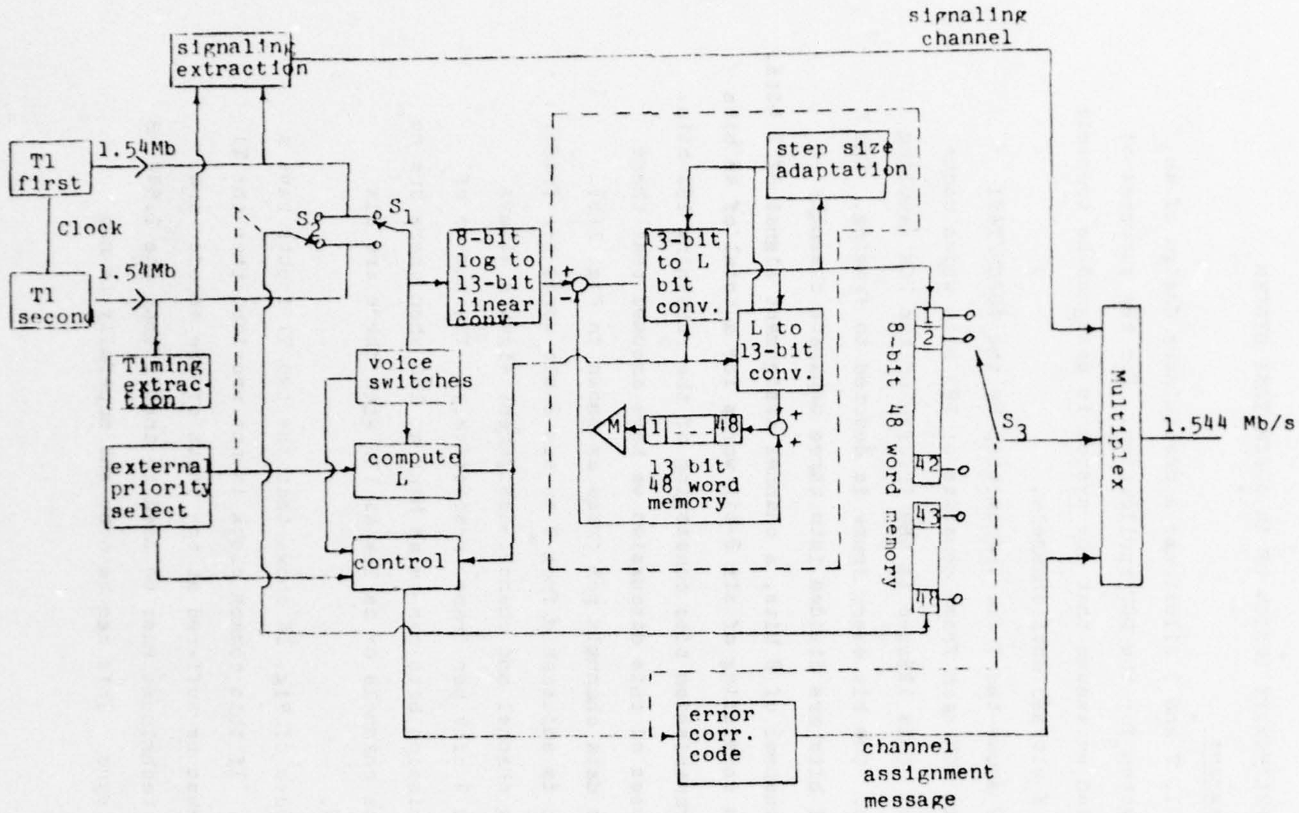


Figure 14. Block diagram of ADPCM/TASI encoder

The "signaling extraction" box removes the signaling information from all speech channels and forms a separate 8-bit per frame signaling channel. The "timing extraction" function determines frame and bit synchronization. Although not shown on the diagram these timing signals are passed to the other parts of the system which need them.

The "external priority selection" function is a set of switches or signals which determine which of the 48 input speech channels should bypass the ADPCM operation. The data channels would be so designated as could any other channel for which the ADPCM operation might prove unsatisfactory. The "external priority selection" controls switch S_2 which feeds the data signals to a data buffer bypassing the ADPCM operation.

Essential to the operation of any TASI-type system are the voice switches, shown in Fig.14, that can recognize the presence or absence of speech in a noisy environment. This task is made quite difficult by the fact that many speech sounds are like noise. All voice switches introduce a certain amount of clipping at the beginning of a speech burst. Speech clips longer than about 50 m sec cause objectionable mutilation of initial fricative, nasal and plosive consonants [14]. The state-of-the-art in voice switch design is currently represented by the voice switch designed for the SPEC system by COMSAT. It operates off of an 8-bit μ=255 PCM bit stream. The starting point for our voice switch will be this COMSAT design which is described in [17]. The number of voice switches on is passed to "compute L" which computes the number of bits L available for each speech channel.

The box marked "control" has as its input the number L, the voice switch information for each speech channel, and the priority information. From these inputs it creates the control signal for switch S_3. This switch selects the channel whose voice switches are on and assigns them a transmission slot L bits wide. This control information contains the channel assignment. It is also encoded using a rate $\frac{1}{2}$ error correction code and multiplexed onto the 1.544 Mb/s transmitted bit stream. The error correction code is needed to protect this control information against errors in the transmission medium.

The "8-bit log to 13-bit linear converter" converts the 8-bit μ=255 PCM input into a 13-bit linear code. This is done so that the ADPCM system (enclosed by a dashed box in Fig. 14) can operate with a linear digital code. The smallest step in an 8-bit μ=255 companded code is equivalent to the smallest step in a 13-bit linear code. This code conversion is relatively straightforward [18].

The quantizer in the ADPCM system is the "13 bit to L bit converter". The feedback loop consists of the "L bit to 13-bit converter", the "13-bit, 48 word memory" and the multiplier M. If we choose M=.875, a reasonable value, this multiplier becomes a simple shift and add. The "step size adaptation" procedure is that recommended by Jayant [3].

The output of the ADPCM system is stored in the "8-bit 48 word memory for speech". The L bit speech samples are held there until the switch S_3 connects them to an appropriate transmission channel. The data samples are held in the "8-bit 6 word memory for data" until the switch S_3 puts them into a data slot.

Figure 15 is a block diagram of the ADPCM/TASI decoder. Its operation is the inverse of the encoder and need not be discussed further.

5.2 Hardware

It is quite difficult to estimate the hardware requirements from the block diagrams of Figs. 14 and 15. The chip count is dependent on the detailed design which has not yet been done. It may be desirable to implement the encoder and/or decoder using programmed controllers. The scale or size of the integrated circuit packages to be used has not yet been determined. Nevertheless, we estimate that the chip count for the encoder to be approximately 150 to 300 chips. For the decoder we expect a chip count from 100 to 250. These estimates include the T1 hardware to be purchased for the "signaling extraction", "timing extraction" and "alarms" functions.

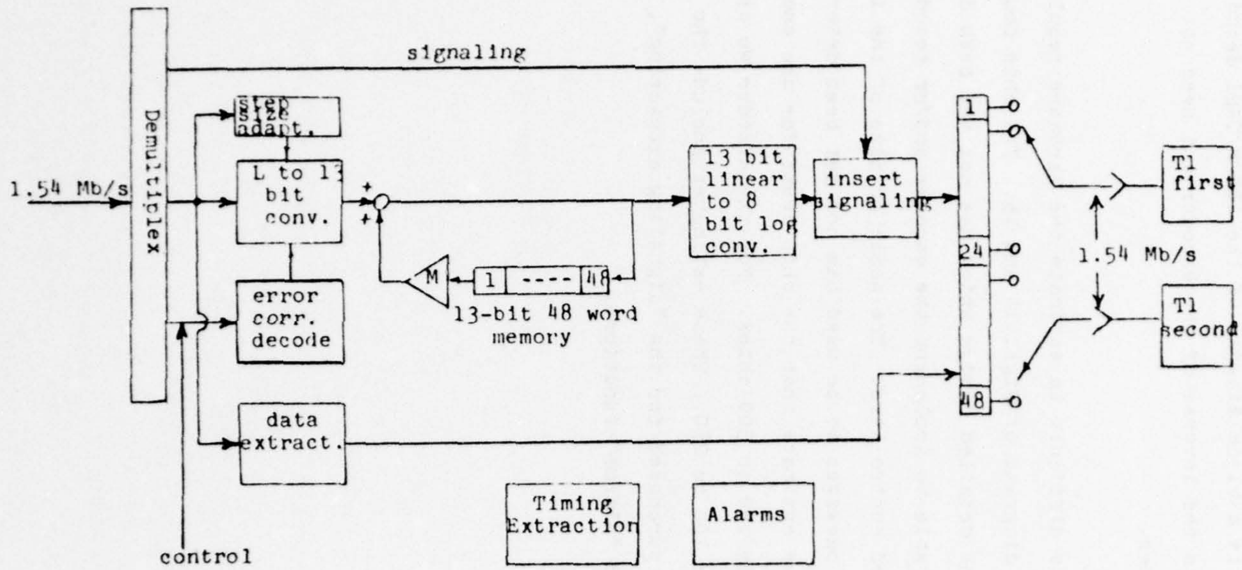


Figure 15. Block diagram of ADPCM/TASI decoder

6. AUDIO TAPE DEMONSTRATION

An audio tape was made to demonstrate the performance of some of the data compression systems described in this report. This audio tape is not furnished with this report. The tape speed is 7-1/2 inches/sec. and the recording is on channel 1. Except for Parts A and J a duty factor of .38 was used.

The contents of the tape are as follows:

Part A: Part A contains the 11 speech segments used as source material for the computer simulations. These sentences are listed in section IV-A of this report.

Part B: 16/0 System - no errors

Male speaker - then female

1. Original
2. SPEC
3. ADPCM/TASI

Part C: 46/2 System - no errors

Male speaker - then female

1. Original
2. SPEC
3. ADPCM/TASI

Part D: 42/6 System - no errors

Male speaker - then female

1. Original
2. SPEC
3. ADPCM/TASI

Part E: Performance of SPEC in the presence of errors - 16/0 system. Male speaker only but with 1 repetition

1. Original
2. $P(e) = 10^{-5}$
3. $P(e) = 10^{-4}$
4. $P(e) = 10^{-3}$
5. $P(e) = 10^{-2}$

Part F: Performance of 5-bit JADPCM in the presence of errors. This is a lower limit on the performance of ADPCM/TASI - 16/0 system. Male speaker with one repetition.

1. Original
2. $P(e) = 10^{-5}$
3. $P(e) = 10^{-4}$
4. $P(e) = 10^{-3}$
5. $P(e) = 10^{-2}$

Part G: Performance of 5-bit JADPCM compared with SPEC with a probability of error of $P(e) = 10^{-4}$. Male speaker with one repetition.

1. Original - no errors
2. SPEC - $P(e) = 10^{-4}$
3. JADPCM - $P(e) = 10^{-4}$

Part H: Performance of 5-bit JADPCM compared with SPEC with a $P(e) = 10^{-3}$. Male speaker with one repetition.

1. Original - no errors
2. SPEC - $P(e) = 10^{-3}$
3. JADPCM - $P(e) = 10^{-3}$

Part I: 3:1 and 4:1 compression with ADPCM/TASI: 3 T1 systems compressed into 1.544 mB and 4 T1 systems compressed into 1.544 mB - no dedicated data. Male speaker with one repetition.

1. Original
2. 3:1 compression (177-bit frame)
3. 4:1 compression (172-bit frame)

Part J: Comparison of ADPCM/TASI with SPEC for 42/6 systems with duty factor of α=.45. Male speaker with one repetition.

1. Original
2. SPEC
3. ADPCM/TASI

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APPENDIX A

Computer Programs

A number of computer programs were written during the course of this study. In this appendix we list the two most important ones:

1. MCDAQ2 - Multiple Channel Differential PCM with an Adaptive Quantizer. This program simulates the ADPCM/TASI system described in this report.
2. SPEC3 - Speech Predictive Encoding Communication system. This program simulates the SPEC system described in this report.

These programs are written in FORTRAN IV G Level, Adage Version. In addition to these programs a large number of support programs, subroutines and input/output procedures were written. If all of this software were included here it would double the size of this report. All of the software is available at North Carolina State University.

```
      PROGRAM SPEC3

C     SIMULATION OF COMSAT SPEC SYSTEM
C     WITH ONE CHANNEL OF REAL SPEECH.
C     THE REMAINING CHANNELS ARE MODELED BY RANDOM
C     EXPONENTIALLY DISTRIBUTED VOICE SWITCH ON AND OFF
C     PERIODS, AND INDEPENDENT SEQUENCES OF UPDATE REQUESTS.
C     QUANTIZATION IS A-LAW, 13 SEGMENT, 8 BIT.

C     R.W.S.  2/25/76
C     VERS. 1, REV. C:
C     SOME NEW STATISTICAL MEASUREMENTS ADDED
C     AND SNR CALCULATIONS MODIFIED FOR LESS TRUNCATION ERROR.

C     VERS. 1, REV. D, 3/2/76
C     FREEZEOUTS & FREEZEOUT ERROR ACCUMULATED AS
C     A FUNCTION OF ACTIVITY

C     VERS. 1, REV. E, 3/29/76
C     ADAPTIVE THRESHOLD ADDED AS AN OPTION

C     VERS. 1, REV. F, 6/1/76...
C     ADAPTIVE THRESHOLD RULES FOR NO = 16 OR 18
C     SOME STATISTICS REMOVED FOR SIMPLICITY

C     VERS. 1, REV. G, 6/1/76...
C     BIT ERROR SIMULATION ADDED

C     VERS. 1, REV. H, 7/23/76...
C     ADAPTIVE THRESHOLD RULE FOR NO = 12 ADDED.

C     SPEC3 IS SPEC1 MODIFIED TO USE FUNCTIONS
C     PCMWD &PCMDC IN PLACE OF Q255 & DQ255
C     JB0,9/7/76

      INTEGER RBUF(1664),WBUF(1664),MSLST(20),CNT(100)
      INTEGER ACTIV(48),ADTHR(48),KZOA(48)
      INTEGER DNT,DSGN
      INTEGER PCMWD,DELTA,PCODE,DCODE,EXPRN
      FRACTION PCMDC,SAMP,QSMP
      LOGICAL END,STATE(100),OFFLG,PFLG,CHIF,ROTFG,ADTFG
      DATA SPLNG/1.5/,ALPHA/.38/,QSCL/1./
      DATA MSLST/20*0/,DELTA/0/,DT/125.E-6/,URNR/134217728./
      DATA PURDS/.87/
```

```
      C   ADAPTIVE THRESHOLD RULE & PROBABILITIES OF EXCEEDING
      C   THRESHOLD VS. THRESHOLD:
            INTEGER ADTHR16(48),ADTHR18(48),ADTHR12(48)
            DIMENSION PURDA16(48),PURDA18(48),PURDA12(48)
            DATA ADTHR12/12*0,1,2,2*3,3*4,2*5,27*6/
            DATA ADTHR16/16*0,2*1,2,4*3,3*4,22*5/
            DATA ADTHR18/18*0,2*1,2*2,4*3,3*4,19*5/
            DATA PURDA12/12*.87,.76,.68,2*.59,3*.53,2*.48,27*.44/
            DATA PURDA16/16*.87,2*.76,.68,4*.59,3*.53,22*.48/
            DATA PURDA18/18*.87,2*.76,2*.68,4*.59,3*.53,19*.48/

      C   BIT ERROR SIMULATION VARIABLES:
            INTEGER TCSB,PCMEP,SAW(48),ESAW(48),EBIT,ESEED
            LOGICAL SBEFG,PBEFG,SERRF,PERRF
            LOGICAL CRDFG
            ESEED=1234567
            PE=.000001
            ISEED=7654321
            CRDFG=.F.

            GO TO 710
      760   WRITE(25,700)
      700   FORMAT('1 CARD READ OPTION')
            CALL YESNO(CRDFG)
      730   IF(.N.CRDFG) GO TO 710
            ESEED=1234567
      770   READ(17,720) NI,NO,MSLST(1),SPG,PE
      720   FORMAT(3I2,F10.5,F15.10)
            IF(PE.EQ.0)  PE=.000001
            IF(NI.EQ.0) GO TO 740
            WRITE(25,750) NI,NO,MSLST(1),SPG,PE
      750   FORMAT('1INPUT=',I2,/,' OUTPUT=',I2,/,' SEGMENT=',I2,/,
          1 ' GAIN=',F6.2,//' PROB ERR=',F10.8)
            GO TO 780


      710   WRITE(25,900)
      900   FORMAT('0NO. OF OFF-HOOK INPUT CHANNELS =')
            READ(10,0)NI
            IF(NI.EQ.0)GO TO 740
            WRITE(25,901)NI
      901   FORMAT('+',33X,I3/'0NO. OF OUTPUT CHANNELS =')
            READ(10,0)NO
            WRITE(25,902)NO
      902   FORMAT('+',25X,I3/'0ENTER MESSAGE SEGMENT LIST:')
            I=1
      10    READ(10,0)J
            WRITE(25,903)J
      903   FORMAT(' ',I4)
            MSLST(I)=J
            I=I+1
            IF(J.GT.0)GO TO 10
            WRITE(25,904)
      904   FORMAT('0OUTPUT FILE DESIRED ?')
            CALL YESNO(OFFLG)
            WRITE(25,913)
      913   FORMAT('0SEED =')
            READ(10,0)J
            IF(J.GT.0)ISEED=J
            WRITE(25,914)ISEED
      914   FORMAT('+',7X,I10/'0INITIALIZE CHANNEL MODELS ?')
            CALL YESNO(CHIF)
            WRITE(25,915)
      915   FORMAT('0STAGGERED PRIORITY ROTATION ?')
            CALL YESNO(ROTFG)
            WRITE(25,916)
      916   FORMAT('0ADAPTIVE THRESHOLD ?')
            CALL YESNO(ADTFG)
```

```
        WRITE(25,905)
  905   FORMAT('0CHANGE OTHER PARAMETERS ?')
        CALL YESNO(PFLG)
        IF(.N.PFLG)GO TO 13
        WRITE(25,906)
  906   FORMAT('0MEAN SPURT LENGTH IN SECS =')
        READ(10,0)SPLNG
        WRITE(25,907)SPLNG
  907   FORMAT('+',28X,F5.2/'0SPEECH ACTIVITY =')
        READ(10,0)ALPHA
        WRITE(25,908)ALPHA
  908   FORMAT('+',18X,F6.3)
        IF(ADTFG)GO TO 13
        WRITE(25,917)
  917   FORMAT('0THRESHOLD =')
        READ(10,0)DELTA
        WRITE(25,911)DELTA
  911   FORMAT('+',12X,I3)
        WRITE(25,918)
  918   FORMAT('0PROBABILITY OF UPDATE REQUEST DURING SPURT =')
        READ(10,0)PURDS
        WRITE(25,912)PURDS
  912   FORMAT('+',45X,F5.2)

  13            WRITE(25,990)
  990           FORMAT('0SPEECH SIGNAL GAIN IN DB =')
                READ(10,0)SPG
        WRITE(25,909)SPG
  909           FORMAT('+',27X,F6.1)
        WRITE(25,920)
  920   FORMAT('0SIMULATE SAW BIT ERRORS ?')
        CALL YESNO(SBEFG)
        WRITE(25,921)
  921   FORMAT('0SIMULATE PCM BIT ERRORS ?')
        CALL YESNO(PBEFG)
  780   IF(.N.(SBEFG.O.PBEFG))GO TO 20
        IF(CRDFG) GO TO 20
        WRITE(25,922)
  922   FORMAT('0PROBABILITY OF ERROR =')
        READ(10,0)PE
        WRITE(25,923)PE
  923   FORMAT('+',24X,E11.3/'0TEST CHANNEL POSITION IN SAW =')
        READ(10,0)NTCPS
        IF(NTCPS.LT.1)NTCPS=1
        IF(NTCPS.GT.48)NTCPS=48
        WRITE(25,924)NTCPS
  924   FORMAT('+',32X,I3/'0ESEED =')
        READ(10,0)I
        IF(I.GT.0)ESEED=I
        WRITE(25,925)ESEED
  925   FORMAT('+',8X,I10)
```

```
C   INITIALIZE:
C   NT = PRIORITY OF TEST CHANNEL (ROTATES)
20      NT=1
C   NP = CHANNEL WITH HIGHEST PRIORITY IN CURRENT FRAME
        NP=NI
C   SET UP ADAPTIVE THRESHOLD RULE:
        IF(NO.EQ.16)GO TO 389
        IF(NO.EQ.12)GO TO 382
C   ASSUME NO = 18:
        DO 387 I=1,48
        ADTHR(I)=ADTHR18(I)
387     KZOA(I)=URNR*PURDA18(I)
        GO TO 381
C   NO = 16:
389     DO 385 I=1,48
        ADTHR(I)=ADTHR16(I)
385     KZOA(I)=URNR*PURDA16(I)
        GO TO 381
C   NO = 12:
382     DO 383 I=1,48
        ADTHR(I)=ADTHR12(I)
383     KZOA(I)=URNR*PURDA12(I)
381     CONTINUE

C   INCREMENT FOR STAGGERED PRIORITY ROTATION:
        DNT=NI-1
        DSGN=1
        CALL OSDR(RBUF,MSLST)
        IF(OFFLG)CALL OSDW(WBUF)
C   SAMPLE COUNTER:
        N=0
C   MEAN SQUARE SIGNAL AND ERROR ACCUMULATORS:
        SQE=0.
        SQS=0.
```

```
C     NO. OF SAWS IN ERROR:
        NSERR=0
C     NO. OF DETECTED SAW ERRORS:
        NDSER=0
C     NO. OF BAD UPDATES:
        NBU=0
C     NO. OF TEST CHANNEL FREEZEOUTS DUE TO SAW ERRORS:
        NFSE=0
C     NO. OF TEST CHANNEL SAMPLES WITH ERRORS:
        NPERR=0
C     INITIALIZE BIT ERROR GENERATOR:
        AURNR=URNR/.9
        BURNR=ALOG(.9)/PE
        NERL=0
        CALL EGEN
C     NO. OF BITS PROCESSED BY BIT ERROR GENERATOR:
        NBCNT=0
C     NO. OF BIT ERRORS:
        NECNT=0
```

```
C   COUNTERS OF TEST CHANNEL FREEZEOUTS AND POSSIBLE FREEZEOUTS:
       NPFO=0
       NFO=0
C   ACTIVITY (NO. OF VOICE SWITCHES ON) HISTOGRAM:
       DO 22 I=1,48
22     ACTIV(I)=0
       IF(PFLG)CALL QSCAL(QSCL)
C   MEAN SPURT LENGTH IN SAMPLES:
       SSPL=SPLNG/DT
C   MEAN GAP LENGTH IN SAMPLES:
       SGPL=(1.-ALPHA)*SSPL/ALPHA
C   THRESHOLD TO GENERATE ZERO-ONE RANDOM VARIABLE
C   WITH P(1)=ALPHA:
       KZO=URNR*ALPHA
C   NUMBER OF SIMULATED CHANNELS:
       NSC=NI-1
C   START SIMULATED CHANNELS WITH STEADY STATE DISTRIBUTION:
       IF(.N.CHIF)GO TO 27
C   ACTIVITY COUNT - START AT 1 SINCE TEST CHANNEL ALWAYS ACTIVE*
       NACTV=1
       DO 25 I=1,NSC
       CALL RAND(ISEED,J)
       IF(J.GT.KZO)GO TO 26
       STATE(I)=.T.
       CNT(I)=EXPRN(.T.)
       NACTV=NACTV+1
       GO TO 25
26     STATE(I)=.F.
       CNT(I)=EXPRN(.F.)
25     CONTINUE
C   TEST CHANNEL SAMPLE IN ENCODER & DECODER FRAME MEMORIES:
27     LTCOD=128
       LRCOD=128
C   CONSTANT FOR UPDATE REQUEST 0-1 RV:
       KZO=URNR*PURDS
```

```
      C  MAIN LOOP:
      C  GET SPEECH SAMPLE:
      50     CALL SGET(J,END)
      C  IF NONE LEFT, EXIT LOOP:
             IF(END)GO TO 2ØØ
      C  COUNT SAMPLES:
             N=N+1
             SAMP=J
             FSAMP=SAMP
      C ATTENUATE SIG BY -SPG DB
             FSAMP=FSAMP*1Ø.**(SPG/2Ø.)
      C FMIN1 ADDED SO THE A GAIN ] Ø WILL NOT CAUSE FRACTIONAL
      C OVERFLOW WHEN FSAMP IS CONVERTED TO SAMP
             FSAMP=AMAX1(-1.,FSAMP)
             SAMP=FMIN1(1.,FSAMP)
      C  ACCUMULATE SIGNAL POWER IN TEST CHANNEL:
             SQS=SQS+FSAMP*FSAMP
      C  QUANTIZE TEST CHANNEL SAMPLE:
             PCODE=PCMWD(SAMP)
      C  FORM MAGNITUDE DIFFERENCE BETWEEN PRESENT AND SAVED CODES:
             DCODE=IABS(PCODE-LTCOD)
      C  IF THRESHOLD IS ADAPTIVE, SET THRESHOLD BASED UPON CURRENT
      C  ACTIVITY & UPDATE REQUEST CONSTANT BASED UPON THRESHOLD:
             IF(.N.ADTFG)GO TO 53
             DELTA=ADTHR(NACTV)
             KZO=KZOA(NACTV)
      C  IF CURRENT PRIORITY OF TEST CHANNEL IS 1 THRU NO. OF OUTPUT
      C  CHANNELS, IT WILL BE UPDATED IF IT WANTS TO BE:
      53     IF(NT.LE.NO)GO TO 7Ø
      C  OTHERWISE, SET UP TO COUNT SIMULATED CHANNELS NEEDING UPDATE:
             I=NP
      C  UPDATE REQUEST COUNTER:
             J=Ø
      C  VOICE SWITCH OF CHANNEL I ON ?
      55     IF(.N.STATE(I))GO TO 57
      C  IF SO, GENERATE Ø-1 RV FOR UPDATE REQUEST:
             CALL RAND(ISEED,K)
             IF(K.LT.KZO)J=J+1
      C  ALL OUTPUT CHANNELS USED YET ?
             IF(J.GE.NO)GO TO 6Ø
      C  NO. GO TO NEXT CHANNEL:
      57     I=I+1
      C  IS THIS NEXT CHANNEL THE TEST CHANNEL ?
             IF(I.LT.NI)GO TO 55
```

```
C   YES. THERE IS AN OUTPUT SLOT FOR THE TEST CHANNEL:
70      IF(DCODE.LE.DELTA)GO TO 65
C   UPDATE TEST CHANNEL IN ENCODER FRAME MEMORY IF
C   THRESHOLD WAS EXCEEDED:
        LTCOD=PCODE
C   SAW BIT ERRORS BEING SIMULATED ?
        IF(.N.SBEFG)GO TO 230
C   IF SO, GENERATE SAW ERROR PATTERN
        CALL SAWEP
C   ANY ERRORS ?
        IF(.N.SERRF)GO TO 230
C   YES.  SIMULATE SAW, ETC.
        TCSB=1
        CALL SAWGN
C   DETECTABLE ERROR ? (CORRECT NUMBER OF 1´S)
        IF(NRS1.NE.NO)GO TO 260
C   NO. TEST CHANNEL BIT IN ERROR ?
        IF(ESAW(NTCPS).EQ.0)GO TO 260
C   NO. POSITION OF TEST CHANNEL BIT IN SEQUENCE OF 1´S
C   THE SAME IN TRANSMITTED & RECEIVED SAWS ?
        KK=0
        DO 205 II=1,NTCPS
205     KK=KK+SAW(II)-ESAW(II)
        IF(KK.EQ.0)GO TO 230
C   NO.  BAD UPDATE:
210     NBU=NBU+1
C   GENERATE RANDOM SAMPLE:
        CALL RAND(ESEED,KK)
        LTCOD=KK.A.377B
        GO TO 280

C   FREEZEOUT DUE TO SAW ERROR:
260     NFSE=NFSE+1
        GO TO 280
```

```
      C  PCM BIT ERRORS BEING SIMULATED ?
      230   IF(.N.PBEFG)GO TO 240
      C  YES.  GENERATE 8 BIT ERROR PATTERN:
            PCMEP=0
            PERRF=.F.
            DO 233 I1=1,8
            PCMEP=.B.PCMEP
            CALL EGEN
            IF(EBIT.EQ.0)GO TO 233
            PERRF=.T.
            PCMEP=PCMEP.O.1
      233   CONTINUE
      C  ANY ERRORS ?
            IF(.N.PERRF)GO TO 240
      C  YES. COUNT PCM ERROR ON TEST CHANNEL:
            NPERR=NPERR+1
      C  PUT CORRUPTED SAMPLE IN DECODER FRAME MEMORY:
            LRCOD=LTCOD.XOR.PCMEP
            GO TO 280

      C  ERROR FREE PCM SAMPLE GOES TO DECODER:
      240   LRCOD=LTCOD
            GO TO 280

      C  ALL OUTPUT CHANNELS FILLED BEFORE TEST CHANNEL WAS REACHED.
      C  COUNT POSSIBLE FREEZEOUT:
      60    NPFO=NPFO+1
      C  IF TEST CHANNEL WANTS UPDATE, COUNT A FREEZEOUT:
            IF(DCODE.GT.DELTA)NFO=NFO+1
      C  SAW BIT ERRORS BEING SIMULATED ?
      65    IF(.N.SBEFG)GO TO 280
      C  YES. GENERATE SAW ERROR PATTERN:
            CALL SAWEP
      C  ANY ERRORS ?
            IF(.N.SERRF)GO TO 280
      C  YES.  SIMULATE A SAW, ETC.
            TCSB=0
            CALL SAWGN
      C  DETECTABLE ERROR ?
            IF(NRS1.NE.NO)GO TO 280
      C  NO. IF TEST CHANNEL BIT IS IN ERROR, DO A BAD UPDATE:
            IF(ESAW(NTCPS).NE.0)GO TO 210
```

```
C   RECONSTRUCT TEST CHANNEL AT SPEC DECODER:
280     QSMP=PCMDC(LRCOD)
C   ACCUMULATE ERROR POWER
        X=FSAMP-QSMP
        SQE=SQE+X*X
C   OUTPUT SAMPLE TO DISK IF DESIRED:
        IF(OFFLG)CALL SPUT(QSMP)
C   ROTATE PRIORITY:
        IF(ROTFG)GO TO 81
C   STANDARD ROTATION SCHEME:
        NT=NT+1
        IF(NT.GT.NI)NT=1
        GO TO 82
C   STAGGERED ROTATION SCHEME:
81      NT=NT+DSGN*DNT
        DNT=DNT-1
        DSGN=-DSGN
        IF(DNT.GE.0)GO TO 82
        NT=1
        DNT=NI-1
        DSGN=1
82      NP=NI+1-NT
C   ACCUMULATE ACTIVITY HISTOGRAM:
        ACTIV(NACTV)=ACTIV(NACTV)+1
C   UPDATE CHANNEL MODELS:
        DO 90 I=1,NSC
C   DECREMENT SPURT/GAP LENGTH COUNTER:
        CNT(I)=CNT(I)-1
C   END OF SPURT/GAP ?
        IF(CNT(I).GT.0)GO TO 90
C   YES. SWITCH TO OPPOSITE STATE:
        STATE(I)=.N.STATE(I)
C   GENERATE EXPONENTIALLY DISTRIBUTED RV FOR LENGTH OF
C   NEXT SPURT/GAP:
        CNT(I)=EXPRN(STATE(I))
C   UPDATE NO. OF CHANNELS ACTIVE:
        NACTV=NACTV-1
        IF(STATE(I))NACTV=NACTV+2
90      CONTINUE
C   LOOP:
        GO TO 50
```

```
C  END OF TEST CHANNEL INPUT FILE:
200    IF(OFFLG)CALL CSDW
       SNR=10.*ALOG10(SQS/SQE)
       SQS=SQS/N

C DBSIG AND DBSIG2 ADDED 9/14/76 TRM
C DBSIG2 IS THE DB OF THE SIG RELATIVE TO A FULL WAVE SINUSOID
       DBSIG=10.*ALOG10(SQS)
       DBSIG2=DBSIG+3.
       X=100.*NFO/N

       Y=100.*NPFO/N
       WRITE(25,950)N,SNR,NFO,X,NPFO,Y
       WRITE(25,952)ACTIV
       WRITE(25,955)ISEED

       WRITE(25,962)DBSIG,DBSIG2
962    FORMAT('0SIGNAL PWR =',F6.1,' DB',/,'0SIGNAL PWR RELATIVE
      1, TO A FULL WAVE SINUSOID =',F6.1,' DB')

       ISEED=7654321
       IF(CRDFG)GO TO 770
       GO TO 760

740    IF(.N.(SBEFG.O.PBEFG))CALL EXIT
       X=NBCNT
       X=NECNT/X
       WRITE(25,960)NBCNT,NECNT,X,NPERR,NSERR,NDSER,NBU,NFSE,ESEED
       CALL EXIT

960    FORMAT('0NO. OF ERROR PATTERN BITS GENERATED =',I10/
      1 ' NO. OF BIT ERRORS =',I10/
      2 ' ACTUAL PROBABILITY OF ERROR =',E12.3/
      3 ' NO. OF TEST CHANNEL SAMPLES WITH BIT ERRORS =',I10/
      4 ' NO. OF SAWS IN ERROR =',I10/
      5 ' NO. OF DETECTED SAW ERRORS =',I10/
      6 ' NO. OF BAD TEST CHANNEL UPDATES =',I10/
      7 ' NO. OF TEST CHANNEL FREEZEOUTS DUE TO SAW ERRORS =',I10/
      8 ' LAST SEED FOR BIT ERRORS =',I10)

950    FORMAT('0',I10,' SAMPLES'/'0S/N =',F7.2,' DB'/
      1 '0NUMBER OF TEST CHANNEL FREEZEOUTS =',I10/
      2 '  PERCENT TEST CHANNEL FREEZEOUTS =',F7.2/
      3 '0NUMBER OF POSSIBLE TEST CHANNEL FREEZEOUTS =',I10/
      4 '  PERCENT POSSIBLE FREEZEOUTS =',F7.2)
952    FORMAT('0ACTIVITY HISTOGRAM:'/(' ',6I10))
955    FORMAT('0LAST SEED =',I11)
```

```
C     EXPONENTIAL RANDOM NUMBER GENERATOR
            INTEGER FUNCTION EXPRN(SPFG)
            LOGICAL SPFG
            CALL RAND(ISEED,JJ)
C     TRUNCATE TO 10*MEAN:
            IF(JJ.LT.6093)JJ=6093
            XX=-ALOG(JJ/URNR)
C     SELECT ONE OF TWO MEANS:
            IF(SPFG)GO TO 850
            EXPRN=SGPL*XX
            RETURN
850         EXPRN=SSPL*XX
            RETURN


            SUBROUTINE YESNO(FLAG)
            LOGICAL FLAG
800         CONTINUE
A                     JPSR              $ICC
A                     ARMD´L;           CHAR: 0
            IF(CHAR.EQ.1RY)GO TO 810
            IF(CHAR.EQ.1RN)GO TO 820
            WRITE(25,899)
            GO TO 800
899         FORMAT(´ ?´)
810         FLAG=.T.
            WRITE(25,898)
            RETURN
898         FORMAT(´ YES´)
820         FLAG=.F.
            WRITE(25,897)
            RETURN
897         FORMAT(´ NO´)
```

```
C   BIT ERROR GENERATOR:
        SUBROUTINE EGEN
C   COUNT BITS PROCESSED:
        NBCNT=NBCNT+1
C   DECREMENT RUN LENGTH TO NEXT ERROR:
        NERL=NERL-1
        EBIT=0
        IF(NERL.GE.0)RETURN
C   TIME FOR AN ERROR:
        EBIT=1
        NECNT=NECNT+1
C   GENERATE EXPONENTIALLY DISTRIBUTED RUN LENGTH
C   TO NEXT ERROR:
        CALL RAND(ESEED,JJJ)
C   TRUNCATE TO PREVENT VERY LONG RUNS & INTEGER OVERFLOW:
        IF(JJJ.LT.5484)JJJ=5484
C   NOTE SCALING TO AVOID ARGUMENTS OF ALOG NEAR 1.
C   AND POSSIBLE ROUNDOFF ERROR ON SHORT RUNS TO
C   NEXT ERROR:
        NERL=BURNR-ALOG(JJJ/AURNR)/PE
        RETURN

C   SAW ERROR PATTERN GENERATOR:
        SUBROUTINE SAWEP
C   CLEAR SAW ERROR FLAG:
        SERRF=.F.
C   GENERATE 48 BIT ERROR PATTERN & SET FLAG
C   IF ANY ERRORS:
        DO 400 III=1,48
        CALL EGEN
        IF(EBIT.NE.0)SERRF=.T.
400     ESAW(III)=EBIT
C   COUNT SAW ERRORS:
        IF(SERRF)NSERR=NSERR+1
        RETURN
```

```
C  SAW SIMULATION:
         SUBROUTINE SAWGN
C  CLEAR OLD SAW:
         DO 410 III=1,48
410      SAW(III)=0
C  PUT A 1 IN TEST CHANNEL POSITION SO NO RANDOM
C  BITS WILL BE PLACED THERE:
         SAW(NTCPS)=1
C  NUMBER OF RANDOM BITS:
         JJ=NO-TCSB
C  GENERATE THIS NUMBER OF BITS IN RANDOM POSITIONS:
         DO 415 III=1,JJ
C  GENERATE A RANDOM NUBER BETWEEN 1 & 48:
416      CALL RAND(ESEED,KK)
         KK=((KK.R.3).A.77B)+1
         IF(KK.GT.48)GO TO 416
C  IF THERE IS ALREADY A BIT THERE, DO IT AGAIN
         IF(SAW(KK).NE.0)GO TO 416
415      SAW(KK)=1
C  SET THE TEST CHANNEL BIT TO WHAT IT SHOULD BE:
         SAW(NTCPS)=TCSB
C  FORM THE RECEIVED SAW & COUNT THE NUMBER OF 1´S IN IT:
         NRS1=0
         DO 420 II=1,48
         KK=SAW(II).XOR.ESAW(II)
         NRS1=NRS1+KK
420      ESAW(II)=KK
C  COUNT A DETECTED ERROR IF NUMBER OF 1´S IS WRONG:
         IF(NRS1.NE.NO)NDSER=NDSER+1
         RETURN

         END
```

```
            PROGRAM MCDAQ2

      C   MULTICHANNEL DIFFERENTIAL ADAPTIVE QUANTIZATION
      C   VARIABLE WORD LENGTH ADPCM
      C   BASED ON JAYANT'S ALGORITHM
      C   WORD LENGTH IS VARIED BASED ON ACTIVITY, I.E.,
      C   NUMBER OF VOICE SWITCHES ON.
      C   ACTIVITY IS SIMULATED AS IN SPEC1 PROGRAM.

      C   2 TO 8 BITS PER SAMPLE
      C   FIXED RATIO STEP SIZES:
      C   64 STEP SIZES WITH A RATIO OF 2**.125
      C   ADAPTATION RULE IS APPROXIMATELY THAT GIVEN BY JAYANT
      C   FOR A 5 BIT SYSTEM WHEN THE WORD LENGTH IS 5 OR GREATER.
      C   FOR SMALLER WORD LENGTHS, THIS STEP SIZE ADJUSTMENT TABLE
      C   IS SUBSAMPLED.

      C   R.W.S.  7/22/76
      C   VERS. 1, REV. A

            FRACTION SAMP,FR,FSD,FRL,FD,DF
            DATA DF/1B/
            INTEGER RBUF(1664:0),WBUF(1664),MSLST(20)
            LOGICAL END
            INTEGER IRTBL(64:0),IDCNT(16:0)
      C   COUNTER INCREMENTS:
            DATA IDCNT /8*-1,2,5,7,9,10,11,13,14/

            LOGICAL STATE(100),CHIF,NWST,CRDRD,MORE,OFILE
            INTEGER CNT(100),WLH(8),EXPRN,A(20),B(20),C(20)
            DIMENSION D(20),E(20),F(20)
            DATA URNR/134217728./,DT/125.E-6/
            ISEED=7654321


            WRITE(25,800)
      800   FORMAT(' MULTIPLE INPUT?')
            CALL YESNO(CRDRD)
            IF(.N.CRDRD) GO TO 810

            OFILE=.F.
            I=1
            WRITE(25,881)
      881   FORMAT(' SEG,INPUT,BITS,SPURT,ACTIV,GAIN ???',//)
      885   READ(10,0) A(I),B(I),C(I),D(I),E(I),F(I)
            WRITE(25,883) A(I),B(I),C(I),D(I),E(I),F(I)
            I=I+1
      883   FORMAT('0',3I5,3F10.5,'      MORE ?',//)
            CALL YESNO(MORE)
            IF(MORE) GO TO 885
            MAX=I-1

            I2=1
      860   IF(I2.GT.MAX) GO TO 850
            MSLST(1)=A(I2)
            NI=B(I2)
            NBPF=C(I2)
            SPLNG=D(I2)
            ALPHA=E(I2)
```

```
        SPG=F(I2)
        ISEED=7654321
        I2=I2+1

        WRITE(25,83Ø) MSLST(1),NI,NBPF,SPLNG,ALPHA,SPG
  83Ø   FORMAT('ISEG   NI   SPEECH BITS  BURST  ACTIV  GAIN',
       1/,1HØ,I3,2X,I3,6X,I3,8X,3F7.3)
        CHIF=.T.
        GO TO 84Ø

  81Ø   WRITE(25,792)
  792   FORMAT('ØOUTPUT FILE??')
        CALL YESNO(OFILE)
        WRITE(25,9Ø3)
  9Ø3   FORMAT('ØENTER MESSAGE SEGMENT LIST:')
        I=1
  25    READ(1Ø,Ø)J
        WRITE(25,9Ø4)J
  9Ø4   FORMAT('  ',I4)
        MSLST(I)=J
        I=I+1
        IF(J.GT.Ø)GO TO 25

  C  INITIALIZE SAMPLE DATA I/O:
  84Ø   CALL OSDR(RBUF,MSLST)
        IF(OFILE) GO TO 842
        CALL OSDW(WBUF)
  842   IF(CRDRD) GO TO 888
```

```
            WRITE(25,92Ø)
      92Ø   FORMAT('ØNO. OF OFF-HOOK INPUT CHANNELS =')
            READ(1Ø,Ø)NI
            WRITE(25,921)NI
      921   FORMAT('+',33X,I3/'ØINITIALIZE CHANNEL MODELS ?')
            CALL YESNO(CHIF)
            WRITE(25,922)
      922   FORMAT('ØSEED =')
            READ(1Ø,Ø)I
            IF(I.GT.Ø)ISEED=I
            WRITE(25,923)ISEED
      923   FORMAT('+',7X,I1Ø/'ØMEAN SPURT LENGTH IN SECS. =')
            READ(1Ø,Ø)SPLNG
            WRITE(25,924)SPLNG
      924   FORMAT('+',28X,F5.2/'ØSPEECH ACTIVITY =')
            READ(1Ø,Ø)ALPHA
            WRITE(25,925)ALPHA
      925   FORMAT('+',18X,F6.3/
           1 'ØNO. OF BITS PER FRAME =')
            READ(1Ø,Ø)NBPF
            WRITE(25,926)NBPF
      926   FORMAT('+',25X,I5,//,'ØSPEECH SIGNAL GAIN IN DB?')
            READ(1Ø,Ø)SPG
            WRITE(25,927) SPG
      927   FORMAT('ØGAIN=',F7.2,'  DB',//)
      888   CONTINUE
      C  INITIALIZE WORD LENGTH HISTOGRAM:
            DO 22 I=1,8
      22    WLH(I)=Ø
      C  MEAN SPURT LENGTH IN SAMPLES:
            SSPL=SPLNG/DT
      C  MEAN GAP LENGTH IN SAMPLES:
            SGPL=(1.-ALPHA)*SSPL/ALPHA
      C  NO. OF SIMULATED CHANNELS:
            NSC=NI-1
      C  THRESHOLD TO GENERATE 0-1 RV WITH P(1)=ALPHA:
            KZO=URNR*ALPHA
```

```
C     START SIMULATED CHANNELS WITH STEADY STATE DISTRIBUTION:
         IF(.N.CHIF)GO TO 4Ø
C     ACTIVITY COUNT - START AT 1 SINCE TEST CHANNEL ALWAYS ACTIVE:
         NACTV=1
         DO 3Ø I=1,NSC
         CALL RAND(ISEED,J)
         IF(J.GT.KZO)GO TO 31
         STATE(I)=.T.
         CNT(I)=EXPRN(.T.)
         NACTV=NACTV+1
         GO TO 3Ø
   31    STATE(I)=.F.
         CNT(I)=EXPRN(.F.)
   3Ø    CONTINUE

C     BUILD QUANTIZER RANGE LOOKUP TABLE:
   4Ø    DX=2.**.125
         X=16383.
         DO 36 I=63,Ø,-1
         IRTBL(I)=X
   36    X=X/DX

C     INITIALIZE SAMPLE COUNTER:
         N=Ø
C     INITIALIZE MEAN SQUARE SIGNAL & ERROR ACCUMULATORS:
         SQS=Ø.
         SQE=Ø.
C     INITIALIZE PREDICTION & QUANTIZER RANGE:
         IY=Ø
         ICNT=Ø
         FR=IRTBL(ICNT)

C     MAIN LOOP:
C     GET NEXT SAMPLE, IF THERE IS ONE:
   5Ø    CALL SGET(IX,END)
         IF(END)GO TO 2ØØ
C     ATTENUATE SIGNAL BY -SPG
         XXX=IX
         XXX=XXX*1Ø.**(SPG/2Ø.)
         IX=XXX
         IIX=IX
         IF(IX.GT.16384) IX=16384
         IF(IX.LT.-16384)IX=-16384
C     COUNT SAMPLES:
         N=N+1
```

```
C   SET WORD LENGTH BASED ON ACTIVITY:
       IBPS=NBPF/NACTV
C   DON'T ALLOW MORE THAN 8 BITS/SAMPLE:
       IF(IBPS.GT.8)IBPS=8
C   ACCUMULATE WORD LENGTH HISTOGRAM:
       WLH(IBPS)=WLH(IBPS)+1
C   SET UP ADPCM PARAMETERS:
       ML=1.L.IBPS
       IMID=ML.R.1
       ML=ML-1
       IRSHF=15-IBPS
       ILSHF=IRSHF-1
       IBPS5=IBPS-5
       NBPS5=-IBPS5

C   FORM PREDICTION ERROR:
       ID=IX-IY
C   QUANTIZE IT:
       FD=ID
C   CLAMP TO QUANTIZER RANGE:
       IF(FD.GT.FR)FD=FR-DF
       IF(FD.LT.-FR)FD=-FR+DF
       FSD=FD/FR
       IQ=(((FSD.R.IRSHF).A.ML)+IMID).A.ML
C   FORM REPRESENTATION LEVEL:
       FRL=((.B.(IQ-IMID))+1).L.ILSHF
       FRL=FRL*FR
C   RECONSTRUCTED SAMPLE = REPRESENTATION LEVEL + PREDICTION:
       IXH=FRL
       IXH=IXH+IY
C   NEXT PREDICTION = .875 * RECONSTRUCTED SAMPLE:
       IY=IXH-(IXH.R.3)
```

```
C   UPDATE STEP SIZE FOR NEXT TIME BASED ON CURRENT LEVEL:
C   FORM MAGNITUDE OF LEVEL STARTING AT 0:
        IQM=IQ
        IF(IQM.LT.IMID)IQM=ML-IQM
        IQM=IQM-IMID
        IF(IBPS5)61,62,63
C   WORD LENGTH .GT. 5 BITS - KEEP 4 MSB'S
63      IQM=IQM.R.IBPS5
        GO TO 62
C   WORD LENGTH .LT. 5 BITS - POSITION IN UPPER
C   PART OF 4 BIT WORD:
61      IQM=IQM.L.NBPS5

62      ICNT=ICNT+IDCNT(IQM)
        IF(ICNT.LT.0)ICNT=0
        IF(ICNT.GT.63)ICNT=63
        FR=IRTBL(ICNT)

C   OUTPUT SAMPLE TO DISK:
        IF(.N.OFILE) GO TO 8000
        CALL SPUT(IXH)

C   ACCUMULATE SIGNAL & ERROR POWERS:
8000    SAMP=IIX
        X=SAMP
        SQS=SQS+X*X
        SAMP=IIX-IXH
        X=SAMP
        SQE=SQE+X*X

C   UPDATE CHANNEL MODELS:
        DO 90 I=1,NSC
C   DECREMENT SPURT/GAP LENGTH COUNTER:
        CNT(I)=CNT(I)-1
C   END OF SPURT/GAP ?
        IF(CNT(I).GT.0)GO TO 90
C   YES. SWITCH TO OPPOSITE STATE:
        NWST=.N.STATE(I)
        STATE(I)=NWST
C   GENERATE LENGTH OF NEXT SPURT/GAP:
        CNT(I)=EXPRN(NWST)
C   UPDATE ACTIVE CHANNEL COUNT:
        NACTV=NACTV-1
        IF(NWST)NACTV=NACTV+2
90      CONTINUE

C   LOOP:
        GO TO 50
```

```
C END OF INPUT FILE:
200   CALL CSDW
      SNR=10.*ALOG10(SQS/SQE)
      SPWR=10.*ALOG10(SQS/N)+3.
      WRITE(25,950)N,SNR,SPWR,(I,WLH(I),I=1,8)
      WRITE(25,951)ISEED
      IF(CRDRD)GO TO 860
850   CALL EXIT
950   FORMAT('0',I10,' SAMPLES'/'0S/N =',F7.2,' DB'/
     1 '0SIGNAL POWER RELATIVE TO A FULL WAVE SINUSOID=
     2,F6.2,' DB',///,
     3 '0WORD LENGTH HISTOGRAM:'/(' ',I2,I10))
951   FORMAT('0LAST SEED =',I11//)
```

```
C  EXPONENTIAL RANDOM NUMBER GENERATOR
       INTEGER FUNCTION EXPRN(SPFG)
       LOGICAL SPFG
       CALL RAND(ISEED,JJ)
C  TRUNCATE TO 10*MEAN:
       IF(JJ.LT.6093)JJ=6093
       XX=-ALOG(JJ/URNR)
C  SELECT ONE OF TWO MEANS:
       IF(SPFG)GO TO 850
       EXPRN=SGPL*XX
       RETURN
850    EXPRN=SSPL*XX
       RETURN


       SUBROUTINE YESNO(FLAG)
       LOGICAL FLAG
800    CONTINUE
A              JPSR,           $ICC
A              ARMD L:         CHAR: 0
       IF(CHAR.EQ.1RY)GO TO 810
       IF(CHAR.EQ.1RN)GO TO 820
       WRITE(25,899)
       GO TO 800
899    FORMAT(' ?')
810    FLAG=.T.
       WRITE(25,898)
       RETURN
898    FORMAT(' YES')
820    FLAG=.F.
       WRITE(25,897)
       RETURN
897    FORMAT(' NO')

       END
```

APPENDIX B

Some Details of the SPEC Simulation

1. Staggered Priority. The SPEC 3 program simulated systems which agreed in every respect except two with the system outlined in [1]*.

The first exception is the priority arrangement. In [1] the channel priority arrangement assigned the highest priority sequentially to channels 1,2,3,··· *i.e.* for the first frame channel 1 has highest priority, for the next frame channel 2 has the highest priority and so on (see Fig. 5 of [1]). We can show that the overload noise power can be reduced if we use the staggered priority rotation scheme 1,64,2,63,3,62,···, Therefore this priority scheme was used in SPEC 3.

2. Threshold settings. The second way in which the SPEC 3 simulations varied from [1] is in the SPEC predictor thresholds. The threshold is defined by a function $\alpha(m)$ where α is the threshold and m is the number of voice switches on (*i.e.*, m is the number of channels which request an update). If α is 0 then every channel requesting an update gets one. If $\alpha=1$ only those channels whose current sample differs from its predicted value by more than 1 gets an update. In general, only those channels whose current sample differs from its predicted value by more than α are assigned a transmission slot.

The following table defines the function $\alpha(m)$ from the SPEC 3 program. Also shown in the table are the probabilities of exceeding the threshold given that the voice switch is on.

*Understanding the next three paragraphs requires a deeper understanding of the SPEC system than is furnished in this report.

The quantities in this table were used in simulating traffic in the SPEC system.

Table B1. Statistics used in generating traffic in the SPEC 3 program.

No. of voice switches on m	Threshold α			Prob. of exceeding threshold given that voice switch is on		
	48/0	46/2	42/6	48/0	46/2	42/6
1-12	0	0	0	.87	.87	.87
13	0	0	1	.87	.87	.76
14	0	0	2	.87	.87	.68
15	0	0	3	.87	.87	.69
16	0	0	3	.87	.87	.59
17	0	1	4	.87	.76	.53
18	0	1	4	.87	.76	.53
19	1	2	4	.76	.68	.53
20	1	3	4	.76	.59	.48
21	2	3	5	.68	.58	.48
22	2	3	5	.68	.58	.44
23	3	3	6	.59	.58	.44
24	3	4	6	.59	.53	.44
25	3	4	6	.59	.53	.44
26	3	4	6	.59	.53	.44
27	4	5	6	.53	.48	.44
28	4	5	6	.53	.48	.44
29	4	5	6	.53	.48	.44
30-48	5	5	6	.48	.48	.44

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METRIC SYSTEM

BASE UNITS:

Quantity	Unit	SI Symbol	Formula
length	metre	m	...
mass	kilogram	kg	...
time	second	s	...
electric current	ampere	A	...
thermodynamic temperature	kelvin	K	...
amount of substance	mole	mol	...
luminous intensity	candela	cd	...

SUPPLEMENTARY UNITS:

plane angle	radian	rad	...
solid angle	steradian	sr	...

DERIVED UNITS:

Acceleration	metre per second squared	...	m/s
activity (of a radioactive source)	disintegration per second	...	(disintegration)/s
angular acceleration	radian per second squared	...	rad/s
angular velocity	radian per second	...	rad/s
area	square metre	...	m
density	kilogram per cubic metre	...	kg/m
electric capacitance	farad	F	A·s/V
electrical conductance	siemens	S	A/V
electric field strength	volt per metre	...	V/m
electric inductance	henry	H	V·s/A
electric potential difference	volt	V	W/A
electric resistance	ohm		V/A
electromotive force	volt	V	W/A
energy	joule	J	N·m
entropy	joule per kelvin	...	J/K
force	newton	N	kg·m/s
frequency	hertz	Hz	(cycle)/s
illuminance	lux	lx	lm/m
luminance	candela per square metre	...	cd/m
luminous flux	lumen	lm	cd·sr
magnetic field strength	ampere per metre	...	A/m
magnetic flux	weber	Wb	V·s
magnetic flux density	tesla	T	Wb/m
magnetomotive force	ampere	A	...
power	watt	W	J/s
pressure	pascal	Pa	N/m
quantity of electricity	coulomb	C	A·s
quantity of heat	joule	J	N·m
radiant intensity	watt per steradian	...	W/sr
specific heat	joule per kilogram-kelvin	...	J/kg·K
stress	pascal	Pa	N/m
thermal conductivity	watt per metre-kelvin	...	W/m·K
velocity	metre per second	...	m/s
viscosity, dynamic	pascal-second	...	Pa·s
viscosity, kinematic	square metre per second	...	m/s
voltage	volt	V	W/A
volume	cubic metre	...	m
wavenumber	reciprocal metre	...	(wave)/m
work	joule	J	N·m

SI PREFIXES:

Multiplication Factors	Prefix	SI Symbol
1 000 000 000 000 = 10^{12}	tera	T
1 000 000 000 = 10^{9}	giga	G
1 000 000 = 10^{6}	mega	M
1 000 = 10^{3}	kilo	k
100 = 10^{2}	hecto*	h
10 = 10^{1}	deka*	da
0.1 = 10^{-1}	deci*	d
0.01 = 10^{-2}	centi*	c
0.001 = 10^{-3}	milli	m
0.000 001 = 10^{-6}	micro	μ
0.000 000 001 = 10^{-9}	nano	n
0.000 000 000 001 = 10^{-12}	pico	p
0.000 000 000 000 001 = 10^{-15}	femto	f
0.000 000 000 000 000 001 = 10^{-18}	atto	a

* To be avoided where possible.

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